TELOS Nx12

Advanced Digital Broadcast Telephone System

USER'S MANUAL

Manual Version 1.6 for software version 1.0 and later
30 July, 2008

Telos Nx12 Manual

© 2007 *TLS Corporation.* Published by Telos Systems/TLS Corporation. All rights reserved.

Trademarks

Telos Systems, the Telos logo, *Zephyr*, *ZephyrExpress*, *Zephyr Xstream*, *Nx12*, *TWOx12*, *Series 2101*, *Desktop Director*, and *Status Symbols* are trademarks of TLS Corporation. All other trademarks are the property of their respective holders.

Notice

All versions, claims of compatibility, trademarks, etc. of hardware and software products not made by Telos mentioned in this manual or accompanying material are informational only. Telos Systems makes no endorsement of any particular product for any purpose, nor claims any responsibility for operation or accuracy.

We reserve the right to make improvements or changes in the products described in this manual, which may affect the product specifications, or to revise the manual without notice.

Warranty

This product is covered by a one year limited warranty, the full text of which is included in this manual.

Updates

The operation of the Nx12 is determined largely by software. We routinely release new versions to add features and fix bugs. Check the Telos web site for the latest. We encourage you to sign-up for the email notification service offered on the site.

Feedback

We welcome feedback on any aspect of the Telos Nx12 or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments.

Service

You must contact Telos before returning any equipment for factory service. We will need the serial number, located on the back of the unit. Telos Systems will issue a Return Authorization number which must be written on the exterior of your shipping container. Please do not include cables or accessories unless specifically requested by the technical support engineer at Telos. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers please contact Telos technical support at +1 (216) 241-7225. All other customers should contact your local representative to make arrangements for service.

We support you...

By email workdwide: support@telos-systems.com

The Telos Web site has plenty of useful information. http://www.telos-systems.com

By phone/Fax in the USA: Customer service is available from 9:30 AM to 6:00 PM USA Eastern Time, Monday through Friday at +1 216.241.7225. Fax: +1 216.241.4103.

By phone/Fax in Europe. Service is available from Telos Europe in Germany at +49 81 61 42 467.

10 9 8 7 6 5 4 3 2 1

Telos Systems USA
2101 Superior Avenue
Cleveland, OH 44114
USA

Telos Europe
Johannisstraße 6
85354 Freising
Germany

Notices and Cautions

This symbol, wherever it appears, alerts you to the presence of uninsulated, dangerous voltage inside the enclosure – voltage which may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions. Read the manual.

CAUTION:
THE INSTALLATION AND SERVICE INSTRUCTIONS IN THIS MANUAL ARE FOR USE BY QUALIFIED PERSONNEL ONLY. TO AVOID ELECTRIC SHOCK, DO NOT PERFORM ANY SERVICING OTHER THAN THAT CONTAINED IN THE OPERATING INSTRUCTIONS UNLESS YOU ARE QUALIFIED TO DO SO. REFER ALL SERVICING TO QUALIFIED PERSONNEL.

WARNING:
TO REDUCE THE RISK OF ELECTRICAL SHOCK, DO NOT EXPOSE THIS PRODUCT TO RAIN OR MOISTURE.

USA CLASS A COMPUTING DEVICE INFORMATION TO USER. WARNING: This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a Class A computing device, as specified by FCC Rules, Part 15, Subpart J, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense. NOTE: Objectionable interference to TV or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. FCC rules require the use of shielded cables.

CANADA WARNING: "This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the Radio Interference Regulations of the Canadian Department of Communications." "Le present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques (de Class A) prescrites dans le reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada."

Table of Contents

We support you... i

1 *Introduction* *1*

Welcome to the Nx12 1

Features & Benefits 1

Why ISDN Makes Sense for Broadcast Talk Show Systems 2

Two-Studio Modes. 4

A Step up in Quality and Flexibility 4

2 *Installation* *5*

2.1 Getting Started 5

2.2 Connections to the Telco circuits 5

2.3 Connecting the Nx12 to Analog Lines 6

2.4 ISDN Technology 101 6

Alphabet Soup: U, S, and NT1 6

Data and Voice 7

Hunt Groups 7

Special Considerations for Euro-ISDN (ETS 300) users 7

Ordering Help 8

2.5 Connecting the Nx12 to ISDN 8

ISDN in USA & Canada: Using the U Interface 8

ISDN Worldwide (outside the USA & Canada): The S/T Interface. 9

2.6 Integration with a PBX 10

Using ISDN 10

Using POTS 10

2.7 Studio Audio Connections 11

Mix-Minus 11

Using a modern broadcast console's mix-minus capability 11

'Making do' with an older console 12

Using a small mixer 13
Using a production-style console 13
Phones and Remotes 13
Input (To Caller). 14
Output (Caller) 16
2.8 Desktop Director Ports 17
Connecting Desktop Directors. 19
Desktop Director Assembly 20
2.9 Ethernet 21
2.10 Parallel GPIO Remote Control 22
2.11 AC (mains) Power 23

3 *Basic Configuration* *25*
3.1 Step-by-Step 25
3.2 Configuration for Telephone Lines. 25
Analog Phone Line Setup 26
ISDN Setup 26
Is the ISDN working? 28
3.3 Audio Input and Output 29
Levels, Levels, Levels. 29
Time to test it out! 30
What now? 31

4 *User Operation* *33*
4.1 Desktop Director 33
Status Symbol Icons 33
Line Button Columns. 35
Hold & Drop 35
Next 35
Talent Mode 35
Block All 36
Dialing Keypad 36
Function Buttons 36
LCD Screen and Associated Buttons 37

Changing Between Producer and Talent Modes 38
Using the Director in Producer Mode. 39
Using the Director in Talent (on-air) Mode 40
4.2 The Call Controller 42
4.3 The Console Director 42

5 *Detailed Configuration & Reference* *43*

First Step... 43
5.1 Level Metering 43
5.2 Status Displays 44
Line Status 44
Desktop Director Status 45
5.3 Using the LCD/Buttons and Web for Configuration 46
The LCD/Button User Interface 46
The Web User Interface. 46
5.4 The Audio Menu. 47
Nominal Input Level 48
Nominal Output Level 48
Meter Mode. 48
Phone Send Adjustment 49
Call Ducking 49
Receive AGC. 49
Noise Gate 50
Receive EQ (Caller Dynamic Digital EQ – DDEQ) 50
Feedback Reduction 50
Mix Minus Mode 50
AES Mode. 53
5.5 The Telco Menu 54
POTS Country 54
Hookflash. 55
Loop Current Checking (Web only) 55
Enable Auto Call Drop (Web only) 55
Ring-in Before CID (Web only) 55

ISDN Protocol 56
SPIDs and MSN/DNs (Web Only). 57
Clock Loss. 59
5.6 The System Menu 59
TCP/IP Setup 59
Contrast (LCD Only) 60
Show Mode (LCD Only). 60
Hold on Select (LCD Only) 60
Audio Loop (LCD Only) 61
Factory Defaults (LCD Only) 61
Reboot (LCD Only) 61
Firmware Update (Web Only) 61
5.7 Show Modes 62
Single-studio Modes. 62
Sharing the Nx12 – Two-studio Modes. 62
Selecting a Show on the Desktop Director and other controllers 64
5.8 Livewire 65
Quality of Service 67
Axia Consoles 67
5.9 Firmware Update 68
5.10 Other Uses of the IP/Ethernet Interface. 68
Call-control Software 68
Telnet 68
5.11 System Signal Flow and Audio Processing 71
Receive (Caller) Audio Processing 71

6 *Technical Data & Troubleshooting* *75*
6.1 Overview. 75
6.2 System Software & Firmware. 75
Viewing Version Information 75
6.3 General Troubleshooting. 75
Thinking about Problem Solving 75
6.4 Troubleshooting ISDN Problems. 77

Status Screen for ISDN troubleshooting 77
Condition: Can't Get ISDN Ready Indication 78
Condition: System says Ready but attempted calls do not complete 79
Condition: Intermittent ISDN Problems 80
6.5 Troubleshooting Desktop Director Problems 81
6.6 Replaceable Modules and Hardware Servicing 81
6.7 ISDN Cause Phrases/Values 82
CLASS 1.1 (normal) 82
CLASS 1.2 (resource unavailable) 84
CLASS 1.3 (service or option not available) 85
CLASS 1.4 (service or option not implemented) 86
CLASS 1.5 (invalid message; e.g. parameter out of range) 87
CLASS 1.6 (protocol error; e.g. unknown message) 87
CLASS 1.7 (interworking class) 89

7 *Specifications* *91*
General 91
Processing Functions 91
Audio 92
Audio Performance 92
Switching Matrix and Conferencing 93
ISDN Telephone Line Connectivity 93
Analog Telephone Line Connectivity 93
Desktop Director Ports 93
Control Ports 93
Two-Studio Modes 94
Four-Column/Four-Hybrid Mode 94
"Euro" Operation Mode 94

8 *Warranty and Application Caution* *97*

A1 *USA SPIDs by Telco* *99*

A2 *Telephone Terminology Guide* *103*

A3 *USA ISDN Ordering Guide* *115*

Dealing with the Phone Company 115

Details, Details 115

FAXABLE ISDN ORDER FORM (for the USA & Canada). 117

A4 *Euro-ISDN Ordering Guide* *123*

Details, Details 123

FAXABLE ISDN ORDER FORM (for Euro ETS 300) 124

1 Introduction

Welcome to the Nx12

The Telos Nx12 combines a 12-line selector and 4 hybrids with advanced signal processing to deliver all you need for your on-air phones. The four hybrids make it ideal for programming where multiple callers are aired simultaneously. It also has the flexibility to serve two studios with two hybrids for each.

At the heart of any broadcast telephone system are the hybrids and associated signal processing. Based on the latest Telos technology, the Nx12 includes the very best broadcast hybrids available.

Depending on the version ordered, the Nx12 can use six ISDN (Integrated Services Digital Network) BRI circuits (with 2 channels per circuit) or 12 analog lines, or a 50/50 mixture of the two types.

The Nx12 has support for up to eight Telos third-generation control surfaces such as our Call Controller and Desktop Director. It also supports control via IP/Ethernet.

Studio equipment can connect with analog, AES3, or Livewire IP audio.

Features & Benefits

Telos products have earned a reputation for ease of use and outstanding performance, and the Nx12 lives up to that reputation. It integrates a suite of audio processing functions to achieve unsurpassed audio quality. Its features include:

- Excellent send-to-receive isolation using the latest generation of Telos' field-tested Adaptive Cancellation technology. Each hybrid automatically and continuously adjusts to phone line conditions, eliminating the hollowness, feedback and distortion that occur when send audio "leaks" into the caller output signal.
- Sophisticated AGC/limiting functions and three-band Digital Dynamic Equalization (DDEQ) ensure controlled caller audio levels and spectral consistency.
- Four high-perfomance adaptive digital hybrids.
- Studio-grade 20-bit audio conversion, combined with noise gating and shaping, extracts the best possible audio quality from real-world telephone lines.
- High-precision sample rate conversion for ISDN circuits.
- Full metering for setup and operation.
- Adjustable Caller Ducking (override).

- Conference linking lets you set up high-quality conferencing between callers with no external equipment needed.
- Caller ID support on both ISDN and POTS lines.
- The Nx12 can serve two studios, each with two hybrids.
- In the two-studio modes, lines can either be shared or exclusive to each studio. In the latter case, there is no possibility for accidental interaction.
- Support for up to eight Desktop Directors.
- Compatibility with Telos Assistant Producer and third-party call control software applications.

Why ISDN Makes Sense for Broadcast Talk Show Systems

The Nx12 works with either traditional analog Telco lines or with ISDN (Integrated Services Digital Network). We recommend ISDN service for telephone connections whenever possible. Contrary to the perceptions of some, ISDN is widely available, cost-effective, and offers many advantages for studio on-air telephone systems. It is yet another example of digital technology enhancing broadcast operations. There is no comparable technology for getting audio to and from the public telephone network. Even compared to VoIP, ISDN is superior, with its lower delay and higher Quality of Service.

To assist you to have a smooth, troublefree installation, we provide plenty of advice right here in this manual, and live support by people with years of ISDN experience. (Many thousands of Telos ISDN interfaces and Zephyrs are deployed around the world!)

Currently, the dial-up telephone network is nearly entirely digital. Only the "last mile" copper connections from the telephone Central Office exchange to the customer's site remain with the archaic (more than a hundred years old…) analog scheme. ISDN offers us a way to link our studios directly into the Telco digital infrastructure without the usual impairments of analog connections.

In all-digital studio facilities, the telephone interface can maintain a digital path all the way using either AES or Livewire networked audio on the studio side

While the application of digital signal processing to the problem of separating announcer and caller audio – pioneered by Telos and used in all of our telephone interfaces – has made a dramatic improvement over analog-only hybrids, using digital phone lines for on-air calls further improves performance for several reasons:

1. **Exceptional send-to-receive isolation.** Traditional analog phone lines multiplex both speech directions in order to use a single pair of wires for a conversation. This causes what's referred to as 'leakage' – when the announcer's audio is present on the hybrid output, creating an annoying 'hollow' or 'tinny' sound. Telos digital adaptive hybrids reduce this problem. But ISDN offers independent and separated signal paths, so our hybrids only have to operate on the far-end analog line, if any. The result is much lower leakage.
2. **Higher send levels**. We don't have to be concerned about regulations designed to prevent crosstalk on analog lines, so we can increase send-to-caller levels.
3. **Lower distortion**. The analog-digital conversion chips used in telephone central offices are poor compared to the converters used in professional audio equipment. Fidelity is not an important consideration when telephone equipment designers choose parts for this function. In a professional interface for studio applications, we can afford to design-in much better converters. Noise-shaping functions permit a larger word-length converter to provide

significantly better distortion and signal-to-noise performance.

4. **Lower noise**. Because they are digital, ISDN lines are not susceptible to induced noise. Analog lines are exposed to a variety of noise and impulse trouble-causers as they snake across town on poles and through your building. Hum is the main problem, given most lines' proximity to pole transformers and AC power wires, but there is also impulse noise from motors, switches, and other sources. Digital lines convey the bits precisely and accurately from the network to your studio equipment without any perturbation – so the audio remains clean. Even when the caller is using an analog phone line, the digital connection on the studio side makes for noticeably lower noise and better overall quality.
5. **Higher gain and reduced feedback during multi-line conferencing.** When conferencing is required on analog circuits, hybrids are needed to separate the two audio paths in order to add gain in each direction. When the gain around the loop exceeds unity, the unpleasant result is feedback. Since the conference path in a studio on-air interface usually includes four AGC functions, the hybrids must be sufficiently good to cover the additional gain that may be dynamically inserted by these AGCs. With digital telephone lines, the hybrid function is more effective – and more reliably so across a variety of calls. That means more gain can be inserted between calls before feedback becomes a problem.
6. **Digital call setup and supervision**. Analog lines use a strange mix of signaling to convey call status. Loop-current drop signals that a caller has disconnected and blasts of 90 Volts at 20 Hz mean someone wants you to answer. Should we be using a mechanism designed to bang a hammer against a metal bell to transmit network status information in the 21st Century?

   ISDN uses a modern digital approach to controlling calls and conveying status information about them. ISDN call set-up times are often only a few 10s of milliseconds, enhancing production of a fast-paced show. Perhaps more importantly, when a caller disconnects while waiting on hold, the ISDN channel communicates this status change instantly. This contrasts with the usual 11-second delay on most analog lines. One of the most common complaints of talk hosts is that they go to a line where they expect a caller to be waiting, only to be met with a blaring, annoying dial tone. The chance of this happening with an ISDN line is nearly zero.

   Another common error is the condition where a talent punches-up a line that looks free, but which actually is just about to begin ringing and connects to a surprised caller. This condition, called glare, results from the delay in the ring signaling, which comes from the nature of the analog line's ringing cadence. ISDN eliminates the ambiguous status period.

   We don't know why the condition of picking up a line which was about to ring and getting an incoming call instead of dial tone is called glare. Perhaps it's because the looks air staff give engineers when this happens too often! If you find out, be sure to let us know.

7. **Better Line Monitoring, Call Setup and Supervision.** Since there is a full-time connection between the central office and the terminal on the D channel, it is possible to detect when a line is not working. On an analog line, one discovers a problem only from a failed attempt to use the line. With ISDN, you can quickly detect when a line is a not working.

Even when ISDN Basic Rate Interface (BRI) lines are not routinely available, ISDN Primary Rate (PRI) lines are. BRIs have 2 channels per line, while PRIs have either 23 or 30. PRIs are well entrenched for use with large corporate PBXs, so are likely to be around for a long time. There are devices on the market that convert from PRI Telco lines to BRI for internal use, so this is a way that the Nx12 could be used long into the future on ISDN.

Two-Studio Modes

The Nx12 can be used with two studios independently. Lines may be independent or shared. Each studio has two hybrids. There are two Progam-on-Hold inputs, one for each studio. Two sets of Telos' Assistant Producer application can be used independently.

Note the following:

- With analog audio in/out, there are only two total inputs and outputs. So the Nx12 must be used in a mode that combines the two hybrids assigned to each studio into one input/output.
- The AES option provides 4 input and 4 output channels total, so all hybrids are accessible with each studio using one AES connection's two channels. Program-on-Hold remains an analog input.
- Livewire provides the full 4 hybrid input channels, 4 hybrid output channels, and 2 Program-on-Hold inputs.

A Step up in Quality and Flexibility

Whether being used with analog or ISDN lines, the Nx12 takes on-air telephone operations beyond the ordinary. Its four hybrids let you have excellent conferencing quality with four simultaneous callers. Or you can serve two studios with one Nx12, with two hybrids for each.

The hybrids in the Nx12 includes long-tail echo cancellation to improve quality on calls originating from VoIP or mobile telephones. Each hybrid is complemented by powerful signal processing functions to extract the best possible quality from real-world phone lines. There are filters, AGCs, multi-band spectral equalizers, and duckers. Sophisticated feedback reduction processing using both acoustic echo cancellation and frequency-shifting enhances performance when an open microphone-to-loudspeaker acoustic path is required.

Telos Desktop Director call controllers offer an easy-to-use but powerful interface for operators. The "Status Symbol" icons provide clear information as to line status with high-contrast LED displays.

Telos Assistant Producer software package gives you what you need for call screening and communication between the producer and talent. Because it connects via Ethernet/IP, it can easily work from remote sites. Because the Telos control protocol is open and published, third-party software has become available offering yet more capability, such as database look-up of caller information and storage of contest results.

Livewire integration is standard. When used with a compatible network and mixing consoles, a single Rj-45 connects all audio and control signals. With the power of the network, new possibilities open such as displaying caller ID on the console's monitor.

2 Installation

2.1 Getting Started

The Nx12 mounts in a 2RU high space in a standard 19" rack. You do not need to pay special attention to cooling or ventilation, as the unit will operate in any environment where the stirred air temperature around the unit is between 0 to 40 degrees Celsius (32 to 104 degrees Fahrenheit) with a relative humidly of 0 to 98% (non-condensing).

Next installation steps are:

- Connect the Telco circuits. Learn about telephone service options.
- Connect audio, via either analog, AES3, or Livewire. Learn about mix-minus.
- Connect your Desktop Directors.
- If needed, connect a PC for Web user interface, either directly or via a LAN (local area network). You can use the front-panel LCD/button interface for most operations, but you will need the Web interface for ISDN and Livewire configuration. And you will need the LAN connection if Telos Assistant Producer or other call screening software is to be used.
- Connect any needed parallel GPIO.

This Installation section covers all of the above. After completing these steps, you will be ready to move on to configuration and testing.

2.2 Connections to the Telco circuits

The Nx12 has two slots for Telco interface modules, each supporting up to 6 lines. Any combination of POTS or ISDN line cards can be used.

If your Nx12 will be "behind" a PBX (private telephone system), see Section 2.6 for more on this integration.

IMPORTANT

Being a broadcast station, you will probably have high call volume. If your PBX has a PRI or T-1 coming from the phone company that will be shared for business operations, it may need to be programmed to reserve some capacity for those calls. One option would be to have the Telco provision (configure) the PRI or T-1 into separate trunk groups for the Business and On-air telephone numbers. You should discuss this with both your local telephone service provider and your PBX provider. You should also inform your Telco that this will be a high volume application so they can be sure that your contests will not interfere with 911 and other emergency services.

2.3 Connecting the Nx12 to Analog Lines

The Nx12 uses plug-in interface modules to attach to POTS (regular, analog loop start, telephone lines). The Country setting in the Telco menu configures the characteristics of the interfaces for your country. If you do not see your country listed you should use CTR21 or USA. These modules have 6 modular jacks. The type of jack will be a 6 position 4 pin RJ-11 style. The connection is to the center two pins of this jack.

Note: Unlike our older products, the Nx12 does not have the loop through provision for POTS lines. The POTS lines used with the Nx12 should be dedicated. The yellow and black leads (pins 2 and 5) of the modular connectors should be disconnected and insulated.

IMPORTANT

When we say POTS lines, we mean just that – plain old analog loop start telephone lines. A rule of thumb – if a line works with an analog modem or fax machine, it will work with the Nx12.

Analog Ground Start lines can be used for incoming calls only. However you will need to set *Loop Check* in the Telco menu to *No* for these lines to be recognized properly.

Be careful not to connect the NX12 to PBX ports intended for proprietary telephones. These sometimes have voltages that could damage the NX12.

2.4 ISDN Technology 101

Since ISDN will be new to some users, we give you here an introduction to the technology here. If you are using analog telephone lines (POTS) or already know ISDN basics, you can skip on to the next section.

ISDN lines come in two varieties: Basic Rate Interface (called BRI, SØ, or ISDN 2 in various parts of the world) and Primary Rate Interface (PRI, S2 or ISDN 30). BRI lines are the kind we normally see in broadcast stations, as these are what are used with MPEG codecs such as the Telos Zephyr and Zephyr Xstream. BRIs have a capability of one or two active 64 kbps channels. Since the Telos Nx12 is used with BRI lines, we will only consider that type here.

Alphabet Soup: U, S, and NT1

An ISDN line from the central office is a single copper pair identical to a normal (unloaded) analog line. When it arrives at the subscriber, it is called the U-interface. It is a two-wire connection, usually via an RJ-11 style modular jack.

The S-interface is at the user side of the Network Termination Type 1 (NT1) device. The NT1 is sometimes generically called a NCTE (Network Channel Terminating Equipment) or, in some countries an NTBA. It is a four-wire connection, via an 8-pin RJ-45 style modular jack. (Sometimes the S-interface is called S/T. There is a subtle distinction between the two, but it is not relevant for our purposes here, and the two may be considered to be the same.)

In the USA & Canada NT1 functionality is usually included in the terminal equipment, and indeed the Nx12 in these countries supports the U-interface. In other parts of the world, the telephone company provides the NT1.

Only one NT1 may be connected to a U-interface. However, as many as eight terminals may be paralleled onto the S bus.

In the USA & Canada a direct connection to the "bare copper" 2-wire U interface is required. Therefore, the USA & Canada interface module includes an integral NT1 and has RJ-11 style jacks. In the rest of the world, the telephone company provides the NT1, and the 4-wire ISDN S-interface will be used with an 8 pin RJ-45 style jack. Normally we ship Nx12s to North America with the U-interface module and to other areas with the S-interface module.

In either case each ISDN interface has three connectors. Since each ISDN connection has two channels, this means each interface module handles 6 'dialtones' or 'lines'.

Data and Voice

ISDN lines may be used for voice signals encoded in standard fashion to allow inter-working with analog telephones, or may be used to transmit digital data streams. The latter mode is used for such applications as high-speed Internet access. It is also the mode used with MPEG codecs, in which case the ISDN line may be carrying voice signals, but is doing so in a format that is not compatible with the analog part of the telephone network. The distinction is made in the automatically conveyed Setup message that begins each call.

Normally, the Telos Nx12 uses only the voice mode, so data capability is not necessary. Often voice costs extra, whereas this is rarely true for data. Of course, you may use a line with both capabilities. Just be sure the BRI circuit supports the Circuit Switched Voice (CSV) capability as well as data.

TIP

If this line may be "borrowed" at times for use with an audio codec, such as the Telos Zephyr, you need to be sure that the Circuit Switched Data (CSD) capability is enabled.

Hunt Groups

The most common configuration for on-air phone system is to have the different phone numbers linked in a "Hunt Group," also called "Rollover Lines," or "Incoming Service Grouping (ISG)." A hunt group allows you to give out a single number to the audience and each call will "hunt" to an unused line. Sometimes one or two numbers will be reserved for a "hot line" or "warm line" in which case those numbers would not be part of the hunt group.

In some cases ISDN lines configured to hunt may deliver all calls to a single DN (phone number). In this case, you must enter this same DN for each of the DNs on each of the hunting channels.

Special Considerations for Euro-ISDN (ETS 300) users

Some Euro-ISDN implementations deactivate BRI circuits that have been idle for more than a given length of time. Since this is a normal occurrence, the Desktop Director and system front panel will not show these lines out of service when this happens. While this will not interfere with normal operation, it can give a false sense of "security" that everything is working when, in fact, the circuit could have stopped working. It is impossible for the Nx12 to distinguish between a deactivated circuit and a malfunctioning circuit until you try to use it. For this reason, if you are using ETS 300, and your phone system is mission critical, you should ask your telephone company to disable this function.

Ordering Help

Appendix 3 to this manual describes the ISDN ordering process for USA/Canada in detail, and Appendix 4 does the same for ETS-300 Euro-ISDN. Both include ready-to-go forms that can be sent directly to your telephone company.

2.5 Connecting the Nx12 to ISDN

The Nx12 uses plug-in modules to attach to ISDN. These ISDN modules have three modular jacks. The type of jack will be either a 6 position RJ-11 style or an 8 position RJ-45 style depending upon whether it is the version for USA & Canada or the rest of the world.

The correspondence between the interface slot to the system line number for is from left to right when viewing the Nx12 from the rear. Therefore, the leftmost jack in slot A would be lines one and two, and the rightmost jack of an ISDN card in slot B would be lines 11 & 12

All ISDN BRI circuits must use the same ISDN protocol setting.

ISDN in USA & Canada: Using the U Interface

Connect the ISDN line from the telephone central office directly to the RJ-11 style U-interface modular jack on the rear panel. Each module has 3 jacks and each jack has an associated LED.

IMPORTANT

Remember that the U interface connects to a long telephone line and can convey lightning surges into the Nx12. Surge protectors intended for analog lines work to protect U-interfaces. We recommend that you install one for each line.

U-interface Status LEDs

The small green LED on the ISDN interface card near each U jack indicate the status of the corresponding ISDN circuit. Rapid flashing (about five times per second) indicates a loss of the ISDN at the lowest (physical) level. If the Nx12 can contact the central office or an active U repeater, the LED will blink slowly – about once per second. The LED will come on solid when handshaking is completed and all is OK. If these LEDs do not light continuously once you have connected your ISDN circuits, you may wish to skip ahead to Section 6.4 for help with troubleshooting the problem.

Nx12 U-interface (RJ-11)

PIN	DESCRIPTION
1	N/C
2	N/C
3	Line R
4	Line T
5	N/C
6	N/C

Note that the polarity of the line connections doesn't matter.

Hot Tip

Normally your telephone company should supply an RJ-11 style jack for the U interface. This is the same 4 or 6-pin plug used for analog lines. However, sometimes you may find an 8-pin RJ-45 style jack–and sometimes you may even be charged extra for it! If you have this situation, an RJ-11 plug can be inserted into the center of an RJ-45 jack to make things work. If you discover that the Telco charged you extra, be sure to request a refund!

ISDN Worldwide (outside the USA & Canada): The S/T Interface

The S-interface version of the Nx12 connects to the NT1 at one of its terminal jacks. The NT1 is sometimes generically called a NCTE (Network Channel Terminating Equipment) or, in some countries an NTBA.

Nx12 S-Interface (RJ-45)

PIN	DESCRIPTION
1	N/C
2	N/C
3	Transmit to network +
4	Receive from network +
5	Receive from network -
6	Transmit to network -
7	N/C
8	N/C

ISDN S-interface cable

Use only an 8-conductor RJ-45 style cable. Unshielded twisted pair category 3 or better cable should be used.

ISDN S-Interface Cable (RJ-45)

PIN	COLOR	DESCRIPTION
1	White/Green	PS3 power +/ground (optional)
2	Green	PS3 power - (optional)
3	White/Orange	Transmit to network +
4	Blue	Receive from network +
5	White/Blue	Receive from network -
6	Orange	Transmit to network -
7	White/Brown	PS 2 power - (optional)*
8	Brown	PS 2 power +/ground (optional)*

TIA 568A Category 5 cable colors shown. Both ends are wired identically – not crossover.
** The NX12 does not use these power connections*

When fully wired, this cable has 4 twisted pairs 'straight through' just as with cables used for Ethernet. Both ends are wired identically. The outside pairs are not required and may be omitted.

2.6 Integration with a PBX

The Nx12 will usually be used independently of any other phone system in your facility. But it is possible to have one or more of the lines come off another phone system. Depending upon the PBX and your preferences, this can be either ISDN BRI or POTS.

You might be able to save money by using a T1/E1 or ISDN PRI as your Telco link and having the PBX generate local ISDN BRI or POTS lines for the Nx12.

Consider having at least one line directly from the Telco to keep your Nx12 up and running even if the PBX or its Telco connection goes down.

Using ISDN

Telos equipment has been used satisfactorily using BRI off the following PBX Switches:

USA and Canada

- Nortel Meridian Option 11, 2 wire (U interface). Protocol ID = 6. [user report]
- Lucent/Avaya Definity, 2 wire (U interface) [user report]
- Avaya IP Office (must use Q.931 mu setting on Telos) [user report]

Outside of the USA

- Siemens HiCom series, 4 wire (S interface) [user report]

ETS300 Euro-ISDN is available from many PBX systems outside the USA and compatibility is generally good.

Unless your PBX is listed above, its support of US/Canada National ISDN-1 is not certain. We recommend you arrange a test for compatibility before committing to the purchase of the PBX ISDN module.

Contact Telos support for the latest information, or to report your experiences with other switches.

Using POTS

Most PBX or VoIP telephone systems can provide analog ports for analog equipment such as modems and fax machines. You will need such ports to connect a POTS Nx12 to a PBX.

With all calls passing through the PBX, calls into the main station number can be transferred over to the on-air system. If the PBX supports 'hook flash transfer' on the analog ports, you can even use this feature of the Nx12 to transfer lines from the on-air system back to the PBX. This can also be done if the PBX and Nx12 share Centrex lines.

A few years ago, we advised that hybrid performance was best when equipment was connected directly to the lines from the Telco. Today, this is not always true. If the PBX is connected to the Telco using digital lines (such as T1, E1, PRI, S2M, etc), performance through a digital PBX will generally be better than direct connections to copper analog circuits. This is particularly so if you are located a significant distance from the Telco central office.

The biggest variable when connecting to a PBX analog port is the quality of the analog port. We have found that many PBX manufacturers offer more than one analog port option. The better port can usually be distinguished by its higher cost, and the fact that it may offer 48 volts rather than 24 volts. The Nx12 will work with either voltage; however experience has shown that the 48-volt version usually has better quality transformers and other components, resulting in better hybrid performance.

The other thing to investigate is whether the PBX offers 'Loop Current Wink' or 'CPC' (Calling Party Control) on the analog ports. Without this supervision, the Nx12 will not detect that a caller on hold has hung up, and your talent will get a recording "please hang up and dial again…" or dial tone when attempting to air such calls.

2.7 Studio Audio Connections

Mix-Minus

The Nx12 must be fed send-to-caller audio that is free of the caller audio, a *mix-minus*. A mix-minus is a mix of all of your audio sources that will be placed on-air (or recorded) except the caller audio – thus the mix-minus designation. The European term M-1 (mix minus one) is perhaps a clearer name. A mix-minus is also sometimes referred to as a *clean feed*. The important thing to remember is that the hybrid must not "chase its tail" – the condition when its output makes its way somehow back to the input.

Hot Tip

Many hybrid installation problems are caused by an inadvertent signal path which creates a loop from the hybrid's output back to its own input. Some consoles allow this when certain control combinations are selected by the user. In some cases, it may be as simple a mistake as assigning the hybrid to whichever bus is feeding the hybrid. This is the first place to look when strange or erratic performance is experienced. The quickest test is to bring up only the hybrid in question on the board and select a line. Dial tone should not appear on the send meter of the hybrid in question.

Using a modern broadcast console's mix-minus capability

Most modern broadcast consoles have provision for multiple mix-minus busses. The best allow selective feeds to the phone system. This is useful since sometimes you want only one microphone feeding the phone, but sometimes you want three or four mics (during the morning show, for instance), and sometimes you want to play some audio piece that callers need to hear and react to such as contest sound effects, etc. Some even provide for separate 'on-air' and 'off line' (recording) telephone modes.

When multiple hybrids are used, each caller will need a separate mix-minus that includes the other hybrids so that callers can hear one another.

Hot Tip

The Axia Element mixing console was designed with today's extensive use of phones and remote codecs in mind. It has all the features described above. It can have as many mix-minuses as faders. It offers an integrated phone controller module that works smoothly with the Nx12 over Ethernet. It has a one-button Record function that does everything needed to record an announcer + phone call off-line. Because it uses Livewire networking, only one RJ-45 is needed to connect the Nx12's four hybrids input and output, two program-on-hold inputs, and all needed control.

'Making do' with an older console

Consoles made before around 1990 rarely had good support for mix-minuses, and almost never for more than one or two. With one of these oldsters, some clever improvisation is going to be needed.

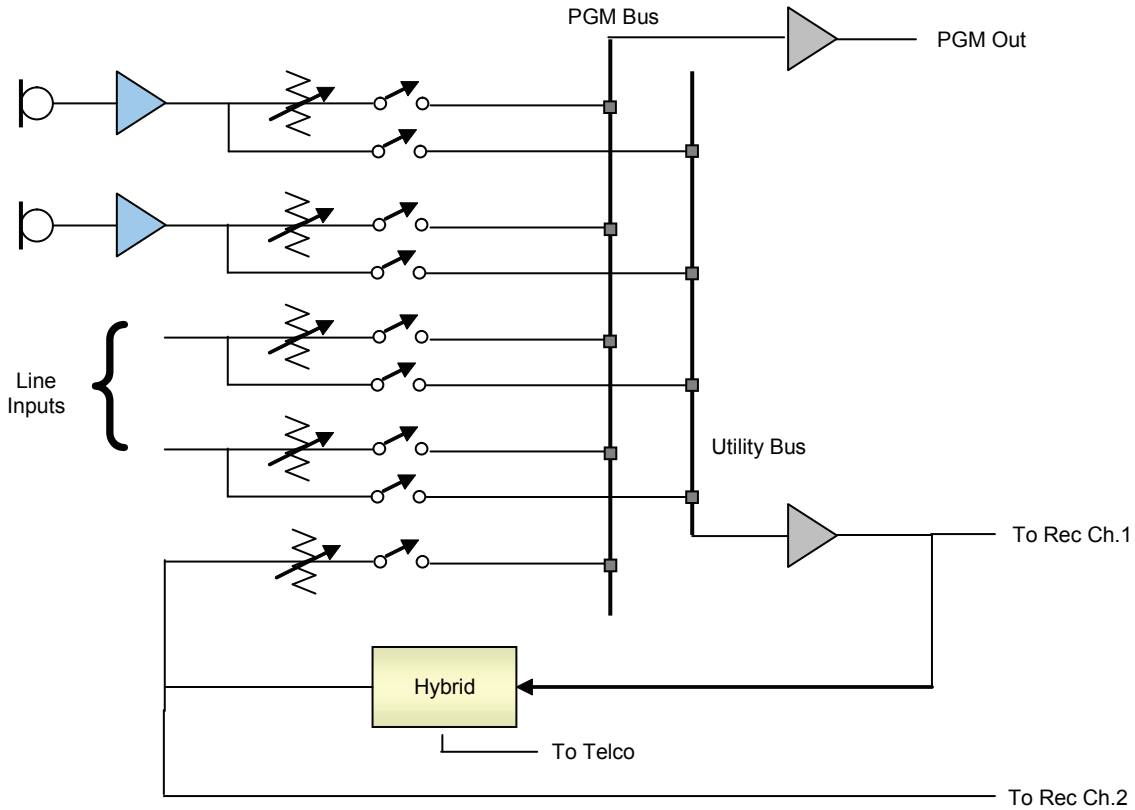
Here we describe a possible scenario that can be used as a starting point for your situation. We assume an older console with Program and Audition as the main busses. There is another bus of some kind that can be adapted for mix-minus application. We'll call this the 'Utility' bus. All sources, including the hybrid, will be assigned to Program, so the audience can hear them, as usual. We will also assign most of these sources to Utility as well, just never the fader with the hybrid's own audio.

This arrangement is flexible, allowing the operator to place any or all sources in Utility for the caller to hear. In our example we have the fortunate case that the console permits the Utility bus to be fed pre-fader, letting the announcer easily use the telephone system for off-air conversations.

A recorder can be attached to the Utility and hybrid outputs to record announcer + phone audio. This is often done as shown here, with each signal to a separate track.

A drawback is the potential for the operator to accidentally put the hybrid in Utility, in which case it is no longer a mix-minus. To avoid this error, the signal path could be permanently disconnected by removing the summing resistors, or some such creative operation.

If no bus is available to feed the Nx12, you could use an external mixer that bridges the microphone inputs to achieve the same effect.

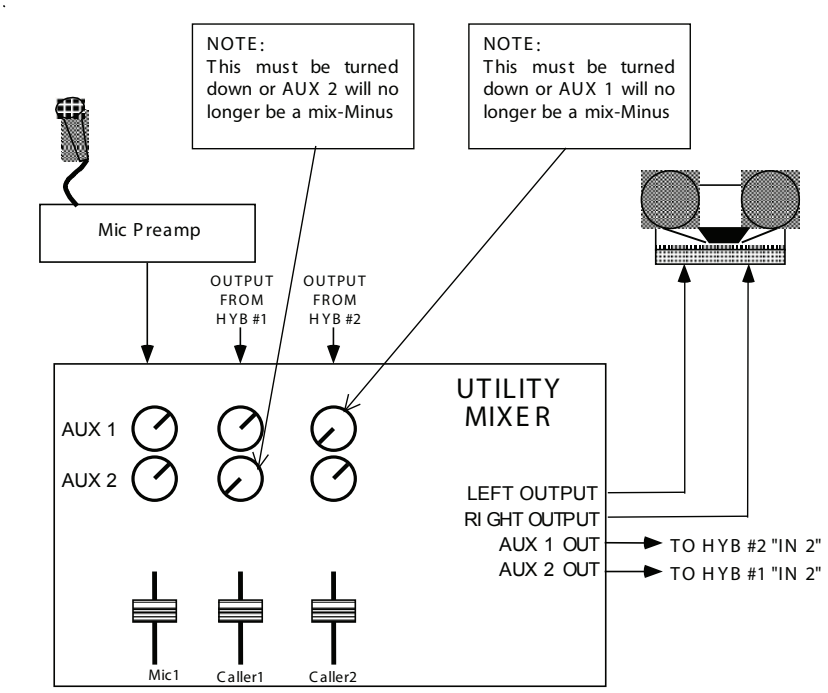


The Nx12 has multiple hybrids and works best if two faders can be assigned to the telephone system with two associated mix-minuses, one for each telephone line column on the Director. This is probably not going to be easy with an older console. But the Nx12 has an option to work with a single external mix-minus by making an internal cross-connection of the hybrids. See Section 5.7 for more on this.

Using a small mixer

In this scenario, a small audio mixer is used to record interviews off the telephone line using a single hybrid. The mixer's main bus is fed to the recording device. Both the microphone and the hybrid will be brought up on the faders so the interview can be recorded.

Most small mixers (such as those made by Mackie) have one or two Aux send busses, so we will use these to feed the telephone system. We will turn up Aux for the microphone but we will make sure it is turned fully off for the each channel that has the corresponding caller audio.



Using a production-style console

The production-style consoles often used for TV audio will have multiple Aux send busses that can be used in a similar way to the small mixer example above. Each hybrid is sent from an Aux bus and everything the caller needs to hear is mixed into that bus, taking care to keep the hybrid itself off the bus.

Phones and Remotes

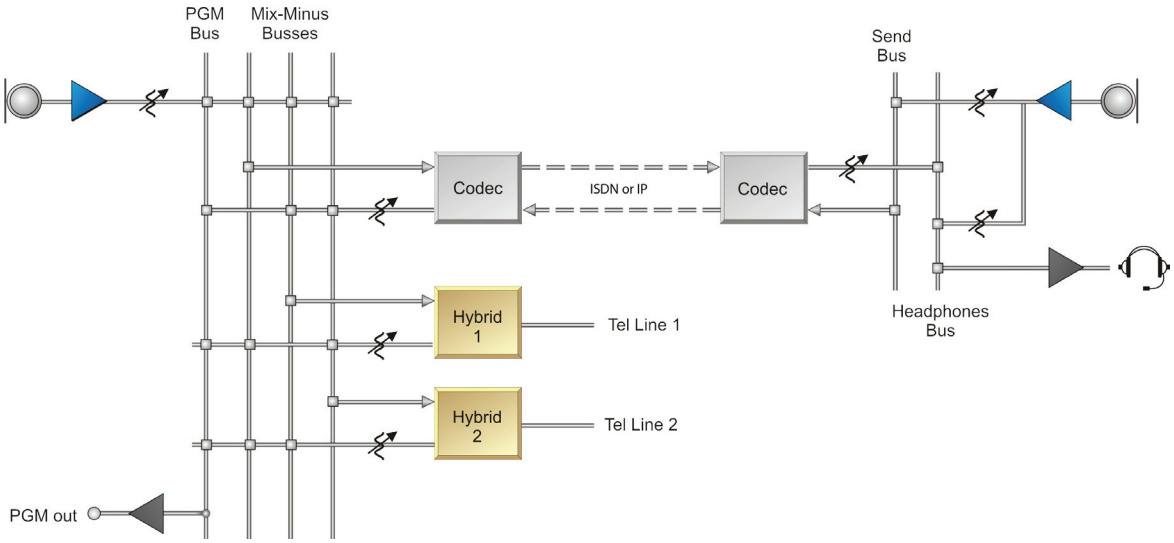
When on remote, to save money and hassle, calls are usually received at the studio, rather than at the remote site. In this situation, caller audio must be fed to the remote talent so that they can hear and respond to callers. Moreover, the callers need to hear the talent. In many cases, the remotes are sufficiently distant that talent cannot monitor the station for the caller feed. Even

if they could, the profanity delay would be a problem, since the talent needs to hear the callers pre-delay.

All perceptual codecs (such as the Telos Zephyr) have too much delay for talent at remote locations to hear themselves via a round-trip loop. Therefore, another mix-minus is required to feed a codec.

The talent hears callers via the codec return path. As before, you feed this return with mix-minus: a mix of everything on the program bus minus the remote audio.

As for the second half of the equation, the callers hear the talent because the remote feed is added to the telephone mix-minus bus. This is no problem if you have a set-up that permits selective assignment to the hybrid mix-minus.



A problem could result from a hybrid with too much leakage and the codec delay. If the hybrid isn't doing a good job of preventing the send audio from leaking to its output, the remote mix-minus is corrupted. Remember, if any of the announcer audio from the remote site is returned, it will be delayed by the digital link, causing an echo effect in the announcer's headphones.

The Telos Nx12 really shows its stuff in this situation. Because it has such good trans-hybrid loss, leakage is not at all likely to be a problem. And should there ever be a problem, you can solve it increasing the amount of ducking. See Sections 5.4 and 5.11.

Input (To Caller)

The Nx12's can be ordered with either analog or AES3 audio connections. A label near the XLR connectors will tell you with which your unit is equipped. In the case of analog there are two inputs. In the AES case there are two physical inputs, corresponding to four signal inputs.

Analog Inputs

The analog inputs have the following characteristics:

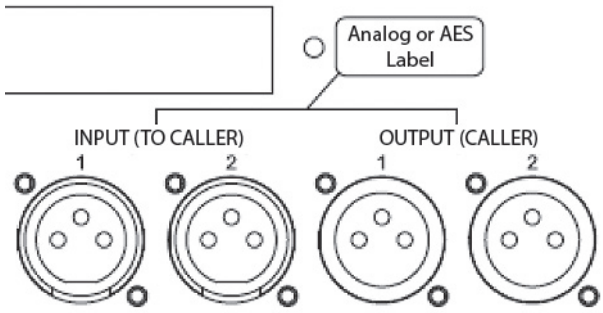
- Active balanced
- Line level: -10dBv to +8dBu nominal level
- Maximum input level (clip point): +24dBu

- Bridging impedance (10k ohms)

Analog Input XLR

PIN	DESCRIPTION
1	Ground
2	Audio +
3	Audio -

The inputs are designed to be sourced from balanced, line level signals. Usually shielded cables



have the shield wire connected only on one end to prevent ground loops. Older equipment with a transformer output stage may need a terminating resistor across pins 2 and 3; consult the manual for your equipment for how to use it with high impedance inputs.

When using unbalanced sources, we recommend using pin 2 and 3 to connect the signal and ground, respectively. Usually pin 1 is not connected in order to avoid ground loops.

The sensitivity of the send inputs are adjusted using Nominal Input Level in the Audio menu. The Input meter indicates the level after the adjustment and before the AGC. See Sections 3.3, 5.1, and 5.4 for additional information on level adjustment.

Note

The Telos Nx12 uses the currently-favored pin-outs for three-pin XLRs. You can easily remember the correct signals when wiring connectors by using the phrase "George Washington Bridge." Pin 1 = G = Ground, Pin 2 = W = "+" = White (typical color in mic cable, if there is no white there will be a red conductor), and Pin 3 = B = "-" = Black.

AES Digital Inputs

These are inputs for AES/EBU format digital audio signals and are only present if the unit was purchased with this option. The AES inputs have the following characteristics:

- AES3 compliant
- 110 Ohms
- Sample-rate converted, so may accept sources at any of the common rates, from 32 to 48 kHz.

AES Input XLR

PIN	DESCRIPTION
1	Ground
2	Signal +
3	Signal -

So that all four hybrids are accessible in spite of the limited rear panel space for connectors, we use one AES input for two hybrids. The relationship is as follows:

- AES 1 Left is to Hybrid A
- AES 1 Right is to Hybrid B
- AES 2 Left is to Hybrid C
- AES 2 Right is to Hybrid D

Most mixing consoles with AES outputs are able to deliver the required split. If not, you might need an external AES router or combiner to provide this input. An alternative could be to use an Nx12 configuration requiring only a single mix-minus (see Mix Minus Modes in Section 5.4). The role of each hybrid depends upon the configured Show Mode, which is described in Section 5.7.

POH (Program on Hold) Inputs

This analog input is for whatever audio you want callers to hear when put on hold, usually your main program feed. This is an XLR female. This signal is processed by an AGC within the Nx12.

Output (Caller)

Analog Outputs

The analog audio outputs have the following characteristics:

- Active differential
- Output level: -10 to +8 dBu, nominal
- Clip point: +24dBu
- Impedance: < 60 Ohms x 2

Analog Output XLR

PIN	DESCRIPTION
1	Ground
2	Audio +
3	Audio -

When feeding unbalanced devices, use pin 2 for the signal and pin 3 for the ground. Do not tie pin 3 to pin 1. (Note that this is different from most devices with active balanced outputs. The Nx12 has a sophisticated output stage that emulates a transformer source.)

AES Digital Outputs

The AES outputs are only present if the AES option was ordered. Characteristics are:

- AES3 compliant

- 110 Ohms
- The output is synced and sample-rate converted to either the frequency of the AES Input 1 signal or an internal 48kHz clock, depending upon a menu selection.

AES3 Output XLR

PIN	DESCRIPTION
1	Ground
2	Signal +
3	Signal -

Since, like the inputs, each AES output has two hybrids, you may have to use a routing switcher, distribution amp, or console routing function to direct the two signals appropriately for your installation. An alternative would be to bridge two AES inputs across each output, which is generally an acceptable practice. Then, the mixing console's input selector would be used to extract the correct hybrid signal for each telephone fader channel.

- AES 1 Left is from Hybrid A
- AES 1 Right is from Hybrid B
- AES 2 Left is from Hybrid C
- AES 2 Right is from Hybrid D

2.8 Desktop Director Ports

At the time this manual was released, the following devices were available to connect to these ports:

- Desktop Director
- Extended Desktop Director
- Console Director
- Call Controller

The Nx12 has four Desktop Director ports. Each supports two devices via simple parallel connection, allowing up to eight total devices. However, the Nx12 can only power four devices. If you have more than four devices, you will need to use the external Desktop Director power supply kit, available from Telos.

(The Extended Director counts as two devices for power purposes, but uses one connection. The Nx12 can control eight Extended Desktop Directors, but will power only one of them.)

It does not matter what port you plug your first four devices into, since all ports support power. You can plug two devices into one port with the Desktop Director power supply kit - it comes with a "Y" cable that allows two devices to be powered by one supply.

Desktop Director

This is the most common control surface. It serves as both a phone and call controller, depending on whether it is set to Talent (on-air controller) or Producer (Screener phone). It has two telephone line columns.

The Desktop Director is suitable for Producer screener mode telephone in all cases.

Extended Desktop Director

In Talent mode, adds two more telephone line columns, making four in total.

Call Controller

The call controller, along with a POTS telephone, emulates the functionality of the Desktop Director. It is suitable for any case where a reduced footprint, reduced complexity, or reduced cost is required.

Console Director

Telos provides a kit that can be used to make a drop-in module for many types of mixing consoles.

Note

The Nx12 can also be controlled over Ethernet using Telos Assistant Producer or other software applications, and by the Axia Element Console's Phone Fader or Call Control modules.

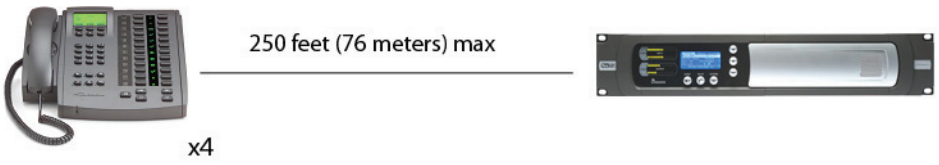
Nx12 Desktop Director Ports (RJ-45)

PIN	DESCRIPTION
1	N/C
2	N/C
3	Transmit to Nx12 +
4	Receive from Nx12 +
5	Receive from Nx12 -
6	Transmit to Nx12 -
7	Power -48VDC
8	Power +

Connecting Desktop Directors

Direct Connection

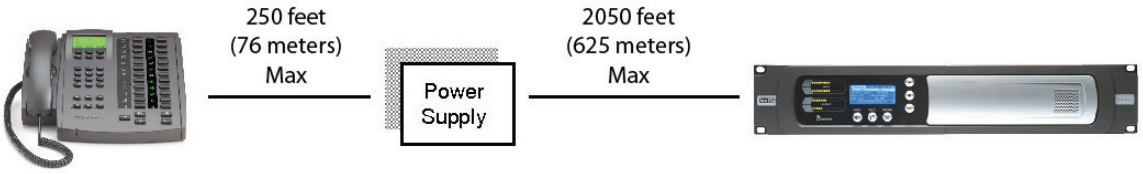
Up to four Desktop Directors can be plugged directly into the Nx12.



Extended Distance with Power Supply

If the Desktop Director must be located more than 250 feet (76 Meters) from the Nx12, you will need to use an external power supply (Telos Desktop Director Power Supply Kit part # 2091-00001).

In that case, your set-up will look like this:

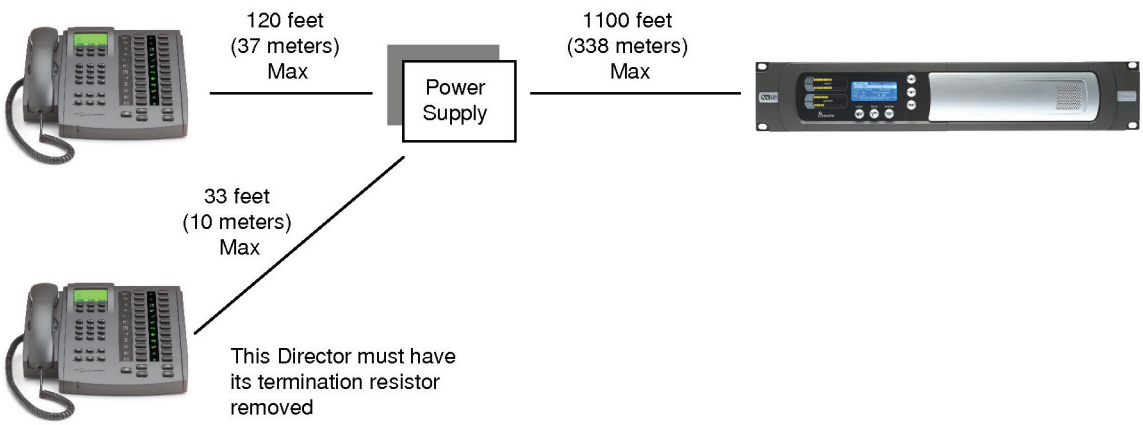


The Desktop Director must be within 2300 feet (700 meters) of the Nx12. The power supply must be 250 feet (76 Meters) or less from the Desktop Director.

Multiple Desktop Directors on a Single Port

Two Desktop Directors can be connected to a single port on the Nx12. You will need to use an external power supply and modular splitter (Telos Desktop Director Power Supply Kit part #2091-00001). You might choose this method to save on cable runs back to an Nx12 located some distance away.

Your set-up will look like this:



To connect two Desktop Directors to one port, a power supply and RJ-45 modular splitter is required.

One of the Directors must have its termination removed. This unterminated Director must be within 33 feet (10 meters) of the power supply.

Desktop Director Termination Removal

Desktop Directors are shipped with the internal termination resistors enabled. When two Directors are used in a single port, the termination on the Director closer to the power supply will need to be disabled. To do so, follow these steps:

1. Disconnect the Desktop Director from the Nx12.
2. Remove the handset and turn over the Director so you can see the bottom of the unit.
3. Remove the round black cover. This can be pried off with a knife blade. Or a miniature screwdriver may be pushed firmly under the lip and then lifted out to pop the cap loose.
4. You will see a black rectangular transformer labeled T1. To the right of T1 locate and remove the jumpers labeled JP1 & JP2. These are small jumper blocks, which can be lifted off with needle-nose pliers.
5. You'll want to save the jumper blocks. You can store each of them on one of the jumper pins.
6. Replace the access cover removed in step 3 by pushing it firmly in place.
7. Reconnect the Desktop Director cable.
8. Check for proper operation

Desktop Director Cable

Each Desktop Director is supplied with a 25-foot cable. If you need another length, you can make your own cable. The Director connection is electrically identical to the ISDN S-interface, so the wiring scheme is the same. A straight-through (not crossover) cable is used between the Nx12 and the Director. TIA 568A or 568B standard Category 5/5e cables may be used. Category 3 cable is acceptable for this wiring. But since category 5e is so common and low-cost, we recommend that you use it instead.

Tip

Power-over-Ethernet and our ISDN-based Director use different connections for power. It is wise to carefully label wiring used with RJ-style connectors to reduce the risk of applying power to something that does not expect it. We suggest that you use different colored cables (or connector boots) on the Desktop Director cables so that you can tell them from your Ethernet cables.

Desktop Director Assembly

Connections

The black "curly cord" plugs into the handset. The other end of this cord should be plugged into the small 4-position/4-pin miniature modular (RJ-22 style) jack near the front left of the unit. This jack is identified with a telephone handset icon and is closest to the front of the Director. The second jack at this location can be used to directly connect a telephone headset and is identified with a headset icon. Route these cables through the small slots to the left so they do not interfere with the unit sitting flat on the table.

When a headset is attached, it will automatically mute the handset microphone to prevent extraneous noise. Disconnecting the "quick-connect" inline connector on the headset cable will re-enable the handset microphone.

Tip

We have tested and recommend the Plantronics model P51-U10P "Supra Polaris noise-canceling unamplified monaural" or model P61-U10P "Supra Polaris noise-canceling unamplified binaural " headsets for direct connection to the Desktop Director. www.plantronics.com. Other headsets can be used; however an external amplifier/interface would be required between Desktop Director's handset jack and the handset to interface to the headset.

Line Designation Strip

The Desktop Director ships with several pre-cut paper designation strips to allow labeling the lines of the Director. Place the paper strip on the director and then cover with the clear plastic protector, inserting the plastic tabs in the slots on the Director to hold it in place. Should you decide to change things later, simply lift one end of the plastic and slide out the strip.

Mounting Options

The Desktop Director legs can be mounted in an alternative vertical orientation. To do so, remove the legs by removing the two Phillips head screws on each leg. The legs can then be mounted in the alternative orientation using the same screws.

The Desktop Director can be mounted on a wall or other vertical surface. First remove the legs by removing the two Phillips head screws on each leg. The unit can then be hung from two screws on 5 inch (127 mm) centers. Care should be taken that the screws do not protrude more than 1/8 inch (3 mm) from the mounting surface.

2.9 Ethernet

The Ethernet 100Base-T port is used for Web-based configuration, connection to Telos Assistant Producer and other software applications, and interface to Livewire networked studio audio components.

The Nx12 fully supports Livewire. For more information on using the Nx12 with a Livewire Audio-over-IP network, see Section 5.8.

IMPORTANT

If you are not using a Livewire-ready network, do not enable Livewire streams or you will flood your Ethernet network. Only when a properly configured Livewire-compatible Ethernet switch is present should Livewire streaming be enabled. The Audio menu includes LW enable/disable as the first item.

The 100Base-T connector has two integrated LED indicators:

- The green Link LED indicates the presence of a live physical connection to a working Ethernet hub (or other device). If the green indicator does not light you should check your network wiring. Absence of the link light could also indicate a hardware failure of the Nx12.
- The amber Activity LED indicates network activity on this segment of the network. Depending on network activity, this indicator may only illuminate occasionally; flickering of this light is normal.

For details on using Web-based configuration, see Section 5.3.

For information on using Telos Assistant Producer and other call screening applications, see Section 5.10.

2.10 Parallel GPIO Remote Control

This port provides inputs to and outputs from the Nx12. Inputs can be connected to logic outputs or switch closures. Outputs can be connected to external device logic inputs.

When any of the two-show modes are used, functions labeled 1 are used for Show 1 and functions labeled 2 are used for Show 2. When show mode is *One Show,* only the functions for 1 are available.

The available functions are:

Record Control Outputs (See Section 4.3)

Ring Mute In – This input will mute the ringers on all of the Desktop Directors connected to this Nx12.

User Out (Delay Dump) – This output will be active (low) when the User button is pressed. It will remain asserted until that button is released.

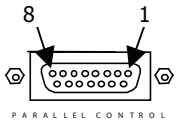
Ringing Out – This output is active (low) whenever the system has one or more lines in the ringing state.

Priority Ringing Out – Active (low) when a line designated as a Priority Line is ringing.

Nx12 GPIO D-Connector

PIN	DESCRIPTION
1	Ground
2	User Out (Dump) 1
3	Ringing Out 1
4	Priority Ringing Out 1

PIN	DESCRIPTION
5	Record Start 1
6	Ring Mute In 1
7	N/C
8	+5VDC (50mA max)
9	Record Stop 1
10	User Out (Dump) 2
11	Ringing Out 2
12	Priority Ringing Out 2
13	Ring Mute In 2
14	(reserved)
15	N/C



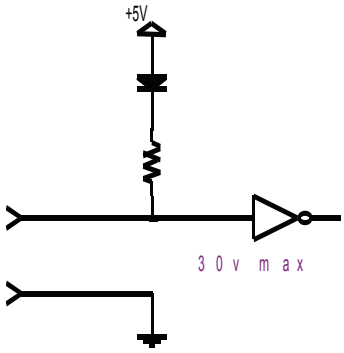
Note for the Curious

The DB- designation for D-Sub connectors is from Cinch Corp's part numbering system. The proper Cinch designation for a 15 pin D-Sub connector is a DA-15p (plug) or DA-15s (socket) not DB-15! (Our thanks to Mike Schweizer, for contributing this tidbit.)

Input Characteristics

- The NX12's inputs are designed to be universal. They accept either a voltage source up to 24VDC, or a closure to ground, which may be provided by switches, relays, or logic outputs. In the latter case either 'totem-pole' or open-collector will work. The inputs are active low.
- A built in 1k Ohm pull up resistor is provided.

Parallel logic input circuit:



Output Characteristics

- Open collector to ground.
- These will require a pull-up resistor to drive TTL-style logic inputs. Most equipment has the pull-up built into the input, but if there is no pull-up, you'll have to add one, connect-

ing it from the output to the +5v source on pin 8. An appropriate value is 2.2kΩ.

- Sink (pull-down) current must be limited to 400mA maximum per output with total output restricted to 1 amp (200mA each output if all five will be used).

If you want to drive a relay, lamp, or high-current LED, you will need an external power source. Connect your external power source negative to Nx12 ground at pin 1 and positive to the relay. The other side of the relay goes to the output pin. (The Nx12's +5v output is fed through a 100Ω resistor and a 50mA fuse, so cannot source much current.)

2.11 AC (mains) Power

The AC receptacle connects mains power to the unit with a standard IEC (International Electrotechnical Committee) power cord and provides an on/off switch. The power supply has a universal AC input, accepting a range from 100 to 240 VAC, 50-60 Hz.

IMPORTANT SAFETY INFORMATION

Surge Protection

Precautions should be taken to prevent damage caused by power surges. Standard line surge protectors can be used to offer some degree of protection. It is the user's responsibility to ensure that protection adequate for their conditions is provided.

Grounding

This equipment is designed to be operated from a power source which includes a third grounding connection in addition to the power leads. Do not defeat this safety feature. In addition to creating a potentially hazardous situation, defeating this safety ground will prevent the internal line noise filter from functioning.

3 Basic Configuration

3.1 Step-by-Step

Here is a step-by-step guide to basic Nx12 configuration for a new installation:

- Enter TCP/IP addresses, if used, for Web interface, remote control, and Livewire audio networking. See Section 5.3.
- Set Input Source for Analog, AES3, or Livewire. See Section 5.8 if you will be using Livewire.
- If you are using Livewire, configure it using the Web interface.
- Connect your mixing console mix-minus(es) to the Nx12's inputs, and the Nx12's outputs to console inputs.
- If you have analog Telco lines, configure for your region. See Sections 3.3 and 5.5.
- If you have ISDN Telco service,
- Are you familiar with ISDN? If not, read Section 2.2.
- Configure ISDN. See Section 3.4.
- Confirm ISDN is working.
- Read-up on Desktop Director Operation. See Section 4.
- Configure the various audio options for processing, etc. See Section 5.4.
- If you will split the Nx12 between two studios, see Section 5.7.

You may use either the Web interface or the front panel LCD/button interface for most configuration operations. The Web may be more convenient since you have a PC's rich interface at your service and you can work by remote.

Some configuration can only be done by Web, such as setting ISDN Directory Numbers and SPIDs and making Livewire channel assignments. See Section 5.3 for detailed information on using these interfaces.

3.2 Configuration for Telephone Lines

The Nx12 has two Telco interface positions, designated A and B. There are three module types, and you can determine which you have by noting the connectors:

- Analog POTS. There will be 6 RJ-11 jacks.
- ISDN U-interface. There will be 3 RJ-11 jacks.

- ISDN S-interface. There will be 3 RJ-45 jacks.

With ISDN, there are two voice channels for each physical connection, so only half as many jacks are needed.

The line list in the LCD Telco menu will correspond to the installed module types, and your configuration entries will vary accordingly.

Analog Phone Line Setup

When POTS lines are used, the only configuration required is to set the Country in the Telco menu. This adjusts the Nx12 for correct impedance, loop current, and signaling. If your country is not listed, try using CTR21 (a European standard) or USA.

ISDN Setup

Determine which physical circuit will represent each line of the system. It might help to fill out the chart below before proceeding. Recall that each BRI has two voice channels, each of which will appear to be a 'line' on the Desktop Director.

SLOT A

		SPIDs (USA/Canada only)	Telephone # DN/MSN
BRI 1	Line 1	_____	_____
	Line 2	_____	_____
BRI 2	Line 3	_____	_____
	Line 4	_____	_____
BRI 3	Line 5	_____	_____
	Line 6	_____	_____

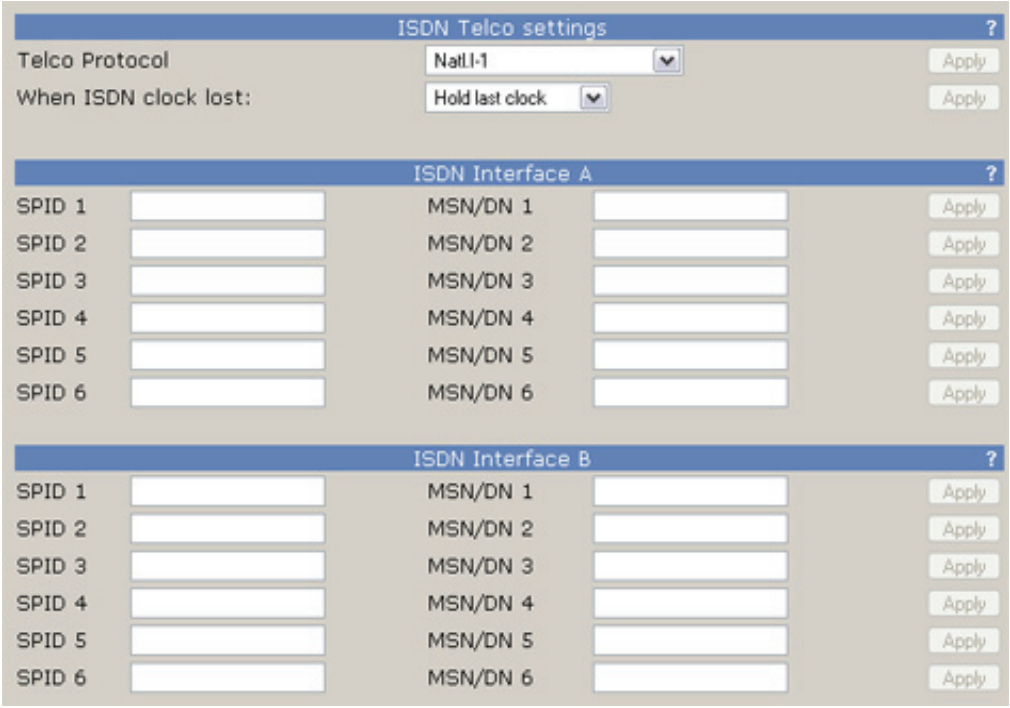
SLOT B

		SPIDs (USA/Canada only)	Telephone # DN/MSN
BRI 4	Line 7	_____	_____
	Line 8	_____	_____
BRI 5	Line 9	_____	_____
	Line 10	_____	_____
BRI 6	Line 11	_____	_____
	Line 12	_____	_____

You will need to use the Web interface for ISDN configuration. The following are step-by-step instructions. See Section 5.3 to learn how to use the Web interface. For additional information and numerous tips on ISDN, see Sections 2.4 and 5.5.

1. Open your Web browser and point it to the Nx12.

2. Click on the Telco menu. With ISDN modules installed in both Telco interface positions, you will see a page similar to the one below.



3. Choose the ISDN Protocol. For the USA and Canada, this will usually be Natl.I-1. For Europe and much of the rest of the world, it will be ETS300.
4. For Natl.I-1, enter the SPIDs for each voice channel given to you by the telephone company. Leave this blank for other protocols.
5. Enter the DN (Directory Number) or MSN for each voice channel. The DNs, sometimes called Listed Directory Numbers or LDNs, are your (usually 7-digit) phone number as listed in the telephone directory. They normally do not include the area code.
6. In some cases, you may need to enter 10 digits for your DNs. If you have difficulties with incoming calls, try this.
7. Click **Apply**, then **Reboot Now** to activate the ISDN circuits.

Tip

Each ISDN Line has a certain ISDN protocol (or 'switch type'), which the phone company should have given you. By far the most common protocols in the USA & Canada are National ISDN-1 & National ISDN-2 in which case you should set the Telco setting to Natl.I-1.

If are in the USA or Canada and you do not know what protocol your line is configured for, try Natl.I-1 first. If that does not work, see Section 5.5 for hints on how to guess your protocol.

More Tips

Incorrectly entered SPIDs are the most common problem users in the USA & Canada have placing ISDN calls. Compare the SPIDs you have entered with the SPIDs provided to you on by the phone company to be certain that you have entered them correctly. Do not add anything to your SPIDs. Note that there are no hyphens, dashes (-), or spaces in SPIDs so the system will not allow you to enter any non-numeric characters. If your installer has included them in the SPID, ignore them.

For a list of known working USA SPIDs by telephone company, see Appendix A.

If you have PTP (AT&T Point-to-Point) or ETS300 (Euro-ISDN) ISDN protocol, you should not enter anything in the SPID fields.

Hot Tip

If you experience problems only with *incoming* calls, check your DNs.

Is the ISDN working?

Successful initialization of a line will be indicated by the idle line state on the Desktop Director. This state is shown with a single dot in the center of the LED icon display being lit.

You can also check the line status on the Nx12 main box LCD. Press any menu button twice and then press the ∇ button to cycle through the status screen. You should see a screen that has A – ISDN B – ISDN at the top. If you do not see this screen, just repeatedly press the ∇or Δ button until you see this screen.

If you do not see READY for each ISDN channel on the Status screen, try rebooting the system from the System/Reboot menu. If you still do not get READY, you will need to check both your ISDN lines and the ISDN configuration settings.

IMPORTANT

Users in the USA & Canada should see READY status for each line whenever the system is idle. On certain ETS-300 ISDN lines, the lines may go from READY to DEACTIVATED after a time interval. This is no cause for alarm and you can proceed with the following steps.

Once you see READY for each channel on the Status screens, you can proceed to make a call and verify that the ISDN lines are, in fact, working. We will start with the simplest case, a call from your line 1 to your line 2, as follows:

- Confirm that the Desktop Director you will be using has power and says Talent at the top of the screen. If it says Producer, follow the directions in Section 4.1 to change to Talent mode.
- Pick up the handset and push each line button in the left column and listen for dial tone. You should see the ⇐ icon as each line is selected.
- Select line 1 and enter the phone number for line 2 using the numeric keypad.
- You should now hear a ringing sound and see the ringing Status Symbol (an expanding square) on line 2. Press the right column line 2 button to answer this call. You should see the ⇒ icon.

- Drop this test call by pressing the right **Drop** button.
- Having completed that first call, you know your lines 1 and 2 are working. (If you are using ISDN, you have now tested the first BRI.)
- Now repeat this test by calling from line 2 to line 3, line 3 to line 4 etc. until all lines have been tested.
- Press the **Drop** button after the final call.

Tip

Many phone lines have per-minute usage charges. Don't forget to drop the call once your testing and configuration are complete.

3.3 Audio Input and Output

The Nx12 can take audio input from analog, AES3 (if the option is installed), or Livewire sources. When the AES module is installed, the XLR connectors are used for AES and analog connections are not possible.

When Livewire inputs are activated, the analog/AES inputs are switched off. However, should LW inputs fail, analog/AES will automatically be switched-on.

On the output side, both analog/AES and Livewire can be simultaneously active.

Levels, Levels, Levels

Ok, we assume that you are able to make a call to yourself. And that you have configured your audio input/output settings. Congratulations, you just have some tweaking to do and your basic installation and configuration will be complete.

- Bring up some audio on the mixing console. Observe the two Input meters. Confirm that audio is present and that levels are OK. Section 5.1 has more information on the Nx12's metering.

Tip

If the levels are too high, hybrid performance will be significantly diminished and leakage may occur.

- First set the Nominal In Level in the Audio menu to a nominal level close to that used in your facility (i.e. –10, 0, +4, or the Euro PPM +9dBu or -9dBfs settings). This control sets the level for analog, AES, and Livewire inputs. Section 5.4 has more.
- You should see the audio from the phone line on the Output LED bargraphs.
- If you have two mix-minuses properly set-up, you should see the audio that is output from hybrid 1 on hybrid 2's input and vice-versa.
- Bring up the fader for Phone 1 on your console.
- Adjust the Nx12's Nominal Out Level in the Audio menu. This sets the level for all outputs. Check that all are providing correct levels to your mixing console.

Time to test it out!

Ok, you made it! Another hour and you can go home for the day! First we will check that the mix-minus for each hybrid is correct. Then we will make a test call.

- Confirm that the Desktop Director you will be using is working and says Talent at the top of the screen. If it says Producer, follow the directions in Section 4.1 to change this Director to Talent mode.
- Next, pick up the handset and press the left column line 1 button to select the line for dialing. You should now hear dial tone on the handset.
- Enter the phone number for a telephone line which has audio present, using the keypad (calling either the time or weather service works well for this).
- Press the left column line 1 button and hang up the handset to place the call on the Phone 1 output.
- Bring up Phone 1 on the console. You should hear the audio from the phone line.
- Look at the Input 1 meter on the Nx12 and verify that this audio is not present. If the phone audio is seen here, you are not feeding a mix-minus and you must fix this problem before you proceed.

IMPORTANT

If you are using one of the mix-minus schemes that require only a single mix-minus you should not have seen audio on either the 1 or 2 bargraphs in the previous test. In that case you can skip the next test. See Sections 2.7 and 5.4 for detailed information about the Nx12's mix minus options.

Now we will verify the second mix-minus.

- Press right column line 2 button to move the call.
- Bring up Phone 2 on the console. You should hear the audio from the phone line.
- Look at the Input 2 meter on the Nx12 and verify that this phone audio is not indicated here. If phone audio is present here, you are not feeding hybrid 2 a mix-minus and you must fix this problem before you proceed.
- Drop the call by pushing the **Drop** button.

Ok, now we just need to test the system as whole. We suggest calling your spouse or partner and reporting that you will be home for dinner after all!

1. Confirm that the Desktop Director you will be using says Talent at the top of the screen. If it says Producer, follow the directions in Section 4.4 to change this Director to Talent mode.
2. Press the left column line 1 button to select that line for dialing. You should now hear dial tone through the mixing console.
3. Enter a phone number to make a call.
4. Bring up Phone 1 on the console. You will hear the audio from the phone line.
5. Put on some headphones, bring up the microphone, and greet the party at the far end. Check that your mix-minus level is OK on the Nx12's Input 1 meter.
6. Your voice should sound normal in your headphones, without any tinny or hollow

artifacts, and the caller should sound clear.

7. Move the call to Phone 2 by pressing the right column line 1.
8. Repeat steps 4 through 6 using lines on the right column.

What now?

Your basic installation is complete. Next you will want to look over other sections of this manual to familiarize yourself with the system and to set up the following parameters:

- Desktop Director operation — See Section 4
- Receive EQ, AGC & Noise Gate — See Section 5.4 and 5.11
- Caller ducking adjustment — See Section 5.4 and 5.11
- Feedback reduction (when using open speakers) — See Section 5.4 and 5.11
- Send EQ, AGC — See Section 5.4 and 5.11
- Mix-minus modes — See Section 5.4
- Show Modes (one Nx12 serving two studios, etc.) — See Section 5.7

4 User Operation

4.1 Desktop Director

The Telos Desktop Director is both a control surface for putting calls on-air and a telephone set for conversations with callers on the studio telephone lines. The Director has two modes of operation – Producer (screener) mode and Talent (on-air) mode.

Tip

Since certain functions work differently in Talent versus Producer mode, it is important to understand that there are two modes, and how to switch between them.

Status Symbol Icons

The Desktop Director uses Status Symbol icons displayed on the LED matrixes located between the two columns of line select buttons. Here is what each icon means:

Line is ready for incoming or outgoing calls

Line is in-use by another studio (in two-studio modes only)

Line is ringing

Call is on hold

Call is On-Air – Column 1

Call is On-Air – Column 2

Call is On-Air & Locked – Column 1

Call is locked and cannot be dropped or placed on hold. To unlock, press the line button to the left of this icon.

Call is On-Air & Locked – Column 2

Call is locked and cannot be dropped or placed on hold. To unlock, press the line button to the right of this icon.

Screened Hold - Call is on hold and has been screened by the Producer

Call is on hold, has been screened, and is Next

Press the **Next** button to put this line on-air.

Block All Mode

Press the X Button to cancel.

Call is on the handset

Call is on the speakerphone

Line is not operational

Line Button Columns

Talent Mode

Each line has 2 buttons, the left column for Caller 1 and the right column for Caller 2. Pressing a line button places the caller on that line on-air using the corresponding hybrid.

Note: When the Nx12 is using a two-studio Show Mode (see Section 5.7), only the left column will be active.

Pressing a line button when a call is already on-air locks the call. To unlock, press the line button a third time.

If the handset is lifted or hands-free is active (the LED next to the hands-free button is illuminated), the left line column will connect the line to either the handset or the Director's internal speaker/mic, rather than to the usual on-air hybrid. The right line column will continue to work as usual. To move a caller on the handset to air, press the line button on either the left or right columns.

Producer Mode

The left row of buttons is used to answer calls on the handset (or headset). Pressing a left column line button while you have a call on the handset (or in hands free mode) locks the caller. To unlock press that Line button a third time.

The right row of buttons is used to place a caller on Screened Hold. For more details on how screened hold works, see the section on the Next button, below.

Hold & Drop

Talent Mode

These affect only the corresponding line button column above them. A locked line will not be affected. All unlocked lines in that column will be dropped.

Producer Mode

The left hold button is normal hold while the right hold button places a caller on "screened hold". For more details on how screened hold works, see the section on the Next button, below. The right hold button may be replaced by with a keycap labeled with a check-mark if a Desktop Director will be used primarily for screening.

Next

Talent Mode

This button will cause the call designated by the call screener as next to be taken on line column 1. Any unlocked caller will be dropped. (It is possible to configure the system to put the caller back hold instead - see 'Direct Hold' option, Section 5.6.)

A producer working as a call screener can place a call on screened hold and the check-mark icon will appear for the line. This indicates that the caller has been checked by the screener, and is ready to go on air.

Calls placed on screened hold in this manner are being queued up from the first selected to the last one selected by the screener. The next caller line icon is represented by a blinking check mark.

If there is no screener to select and order calls, the line which has been ringing the longest will be selected instead. If no line is ringing, the line on hold the longest will be selected.

Note that if the handset is off-hook, the caller will be directed to the handset, using the same priorities outlined above.

Producer Mode

The **Next** button will select the line which has been ringing longest. If no line is ringing, it selects the line that has been on (normal) hold longest.

When **Next** is pressed, any unlocked caller will be dropped. It is also possible to configure the system to put the caller back hold instead. See the 'Direct Hold' feature, Section 5.6.

Block All

Pressing this key will clear all lines except lines on-air, on the handset, on hold and 'priority' lines such as the studio hotline. This allows you to prevent early callers from getting in on contests until after the DJ has made the announcement and turned off block all.

Ringing lines, other than lines designated priority, will be dropped and blocked from accepting any calls. For ISDN lines, the caller may get a busy signal, or they may continue to hear ringing. For analog lines, the ringing calls will be connected to a silent audio channel. Further calls will get a busy signal.

Pressing **Block All** again will release the lines and allow incoming calls.

Dialing Keypad

These are used for dialing out, which can be on the handset, on the speakerphone, or on-air. They remain active during a call for voice mail and other services.

Function Buttons

Speakerphone Button

This button places a call that is on the handset onto this Desktop Director's built-in speaker-phone. If the handset is on-hook this button places the next line selected on the left column on the speaker phone.

Mute Button

This button mutes the handset microphone and hands free microphone on the Desktop Director. The LED beside this button will light to indicate that these microphones are muted. Pressing this button again will cancel the muting function.

Hook flash/Transfer

This button performs a hook flash (briefly interrupts loop current) on POTS lines. Depending on the telephone lines used, this key may serve to activate features such as transfer or 3-Way Calling. This button does nothing on ISDN lines.

User Assigned Control Button ?#@!

This button controls an output on the parallel control port. This output can control equipment external to the Nx12 such as the "delay dump" input on an obscenity delay unit.

Record Control

This is intended to control an external recording device. It operates only if connected.

Pressing the button once when nobody is on air, arms the recording feature (LED blinking).

Once armed, taking a call starts the recorder automatically (LED solid). When the call is dropped, the recorder will be stopped and the system will re enter the "armed" state.

Pressing the button when a call is already on air starts the recorder (LED solid).

While recording (LED solid), pressing the button stops the recorder and puts the system back in the armed (LED blinking).

When in armed state (LED blinking), pressing the button cancels the recording feature.

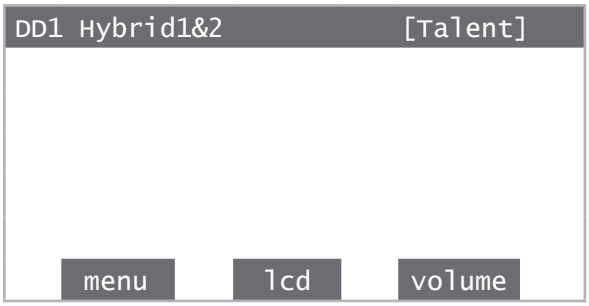
When the Nx12 is used in any of the two-studio modes, the record logic will only operate from Desktop Directors accessing SHOW1. The **Record** button is disabled for any Director connected to SHOW2.

LCD Screen and Associated Buttons

The LCD is used for general information and for configuring the Desktop Director.

Items can be selected and changed using the six buttons immediately below the LCD screen. The buttons in the row immediately below the LCD screen are "soft keys" and their function will be indicated on the bottom row of the screen.

Here is the main, default screen:



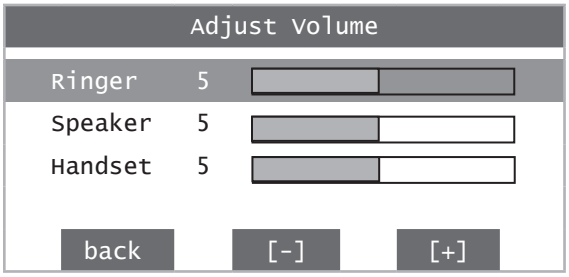
Menu Navigation

The LCD screen and menu navigation buttons are used to adjust volume, LCD screen backlight and contrast, and to change configuration items. The function of the three soft keys, immediately below the LCD screen will vary depending on context and will always be displayed on the screen. In most cases, the left soft key offers a **back** function. This takes you back to the main screen without saving unconfirmed changes (except for in the Volume and LCD screens, where changes are effected immediately).

The Δ & ∇ buttons scroll through the options in a menu list, and the **Select** button chooses a menu item or confirms an entry.

Volume Menus

Pressing the soft key button below the word **Volume** will display the following screen:



Press Δ or ∇ to choose which level you want to adjust. Pressing [-] will decrease, while pressing [+] will increase the volume level for the item displayed. The change will take place immediately. Pressing the soft key button under the word **back** or the **Select** button will return you to the main screen. The following volume levels can each be individually adjusted:

- **Ringer** – Adjusts the volume of the ringer. Note that the ringer will usually be turned off whenever a microphone channel is on.
- **Speaker** – Adjusts the level of the Desktop Director's loudspeaker.
- **Handset** – Sets the level of the caller on the handset. If you are using a headset attached to the Desktop Director, this will adjust the level in the headset as well.

Other Menu options

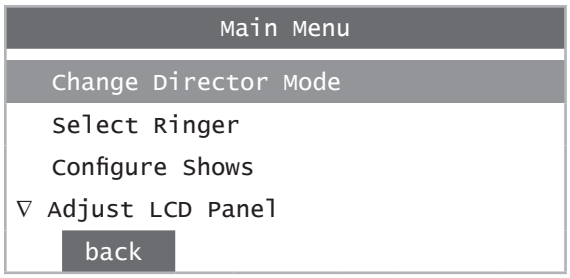
After selecting **menu** from the main LCD screen, the following are offered:

- **Select Ringer** – It is possible to change the ringer tone on each Desktop director. There are four different choices. Press **Select** to apply your changes.
- **Configure Shows** – This gives access to the programing functions for two-studio modes, where each pair of hybrids is assigned to one of two studios. See Section 5.6.
- **Adjust LCD Panel** – Gives access to the same menu as the LCD button on the main screen.
- **Diagnostics** – This gives a status report of the connection between the Desktop Director and the Nx12 unit to which it is connected. Two values are reported:
  - Near End: the number of lost or corrupted data packets in the Nx12 to Desktop Director direction since this Director was powered on.
  - Far End: the number of lost or corrupted data packets in the Desktop Director to Nx12 direction since this Director was powered on.
- **Version** – Reports the currently running Nx12 software version number and the currently running Desktop Director software version number.
- **About** – General information.

Changing Between Producer and Talent Modes

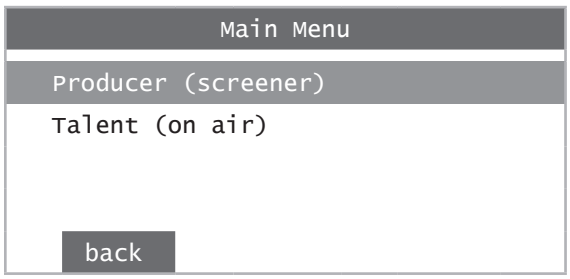
A Desktop Director can be in either Producer or Talent mode. This can be changed at any time independently for each Director in the system. Producer mode can be thought of as 'screener phone' mode while Talent mode can be thought of as 'on-air' mode. To change between Producer and Talent modes:

1. On the main screen, press the soft key button below the word **Menu**. You should now see a screen that looks similar to the following:



The words **Change Director Mode** should be highlighted.

2. Now press the **Select** button. You will now see a screen like this:



3. Press ∇ or Δ to change the setting.
4. Press **Select** to confirm your choice or press the soft key below the word **back** to return to the main menu without any change taking effect.

Using the Director in Producer Mode

The first text line on the LCD tells you which mode you are in. You will see `[Producer]` if you are in this operation mode.

Answering calls

You use the left column of line buttons to answer calls on the handset (or headset).

The **Next** button will select the line which has been ringing longest. If no line is ringing, it selects the line which has been on normal hold longest. When the **Next** button is pressed, any lines not locked will be dropped (or put on hold, if your system is configured for "direct hold" operation).

Normal Hold

The left hold button places a call on normal hold.

Screened Hold

The right hold button places a call on "screened hold". (This button's keycap may be labeled with a check mark if the Desktop Director will be used primarily for screening.)

Designating the Next Call

Normally, when the talent presses his **Next** button, the call waiting on hold for the longest time is selected to air. However, the Producer can force any line to the top of the queue. You use the right column of line buttons to select which call is designated as next - the call that goes on-air when the talent presses his **Next** button. The line status icon will change to the blinking check mark.

Hot Tip

You can use this feature to re-order the next queue at any time. Simply press the right line buttons for the calls in reverse order from the order you wish to air them. Press the right line button first for the call to be aired last, then the button for the next line to be aired before that and so on pressing the right line button for the call to be aired next last.

Block All

Pressing this key will clear all lines except lines on-air, on the handset, on hold and "priority" lines such as the studio hotline. This allows you to prevent early callers from getting in on contests, until after the DJ has made the announcement and turned off Block All.

Ringing lines other than priority lines will be dropped and then be blocked from accepting any calls. For ISDN lines, the caller may get a busy signal, or they may continue to hear ringing. For analog lines, the ringing calls will be connected to a silent audio channel. Further calls will get a busy signal.

Pressing the **Block All** button again will release the lines and allow incoming calls.

Conferencing callers on the handset/headset

The usual operation of the line button is such that selecting a line will cause an existing line in use to be dropped (or placed on hold if your system is programmed to do so). To conference multiple callers, you can use the lock function: With the first call on the handset, press the left line button to lock that call. Now you can select another line without losing the first call. If you wish to add a third party, lock the second line by pressing the line button for the second caller. You can add additional lines in this manner. To unlock a line, press the line button for that line.

At any time, you can drop all unlocked calls by pressing **Drop**, or you can place all unlocked calls on hold or screened hold by pressing the appropriate **Hold** button.

Using the Director in Talent (on-air) Mode

The first text line on the LCD tells you which mode you are in. You will see `[Talent]` if you are in this operation mode.

Answering calls

Each line has two buttons, the left column for Caller 1 (your console fader might be labeled "Phone 1" or something else) and right column for Caller 2. Pressing a line button places the caller on that line on-air on the corresponding fader. When you select the line, the Status Symbol icon will change to an arrow pointing to the line column you used.

Note: In some cases, when an Nx12 system is shared between two studios, only the left column is used.

Pressing a line button when that line is already on-air locks the caller. You can use this capability to prevent accidentally losing a guest. To unlock, press that line button a third time.

If the line has already been screened, the Screened symbol - a check mark - will be displayed. If the caller has not been screened, the hold symbol or ringing symbol will be displayed.

Drop and Hold

A call may be dropped or placed on hold by using the buttons below the line columns. Also, if you select another line on the same column, any other active non-locked lines will be automatically dropped.

Next

This button will take the call designated as next on the left line column. If no call is currently designated next, the call which has been on Screened Hold the longest will be taken. If a screener is not used, the line which has been ringing longest will be taken. If no line is ringing, the line on hold the longest will be taken.

Block All

Pressing this button will clear all lines except lines on-air, on the handset, on hold and "priority" lines such as the studio hotline. Calls ringing-in on lines other than priority lines will be dropped and the lines will then be blocked from accepting any calls. Pressing **Block All** again will release the lines and allow incoming calls.

This allows you to prevent early callers from getting in on contests until after the announcement is made and Block All is switched off.

Hot Tip

As you may know, there have been several lawsuits in the USA regarding to the fairness of radio contests. By consistently activating the Block All feature before announcing a contest you will reduce the liability for yourself and the station.

Conferencing multiple callers on-air using call locking

If you only need two calls at once, you can just use the two line columns, with one call on each. However, if you need more than two, or if your system is serving two studios and you only have one active line column, you can use the lock function for conferencing.

- Press an already on-air line button a second time to lock the line. The status symbol will change to display the locked icon.
- If you take another line on the same column, any locked lines will remain on-air.
- When a locked line button is pressed, the call reverts to normal and the left or right-pointing arrow icon is displayed.
- The **Drop** and **Hold** buttons have no effect on a line in locked status, but apply to any other active calls.

Using the lock feature, you can organize any combination of calls to be conferenced. You can also drop or add calls to an active conference group.

Locking VIP calls

As described above, any line on the air can be locked by hitting the line button a second time. This is a handy feature for use with VIP guest callers – by locking your guest you cannot drop the call by accident, no matter how crazy things get in the studio!

Using the Director as a Telephone

Making a call using the handset. Pick up the handset, select a line using the left column and dial. If you wish to place this caller on air, you can press a left or right line column button. If you hang-up the handset, the caller will be lost unless you have placed them on hold or on-air.

Using the studio microphone and cue speaker. This is identical to placing callers on-air - just be sure that your microphone and phone faders are in cue, not program! If you experience

feedback, lower the cue speaker volume.

Making calls using the built-in speakerphone. Press the **Speakerphone** button, select a line using the left column and dial. If you wish to place this caller on air, you can press a left or right column line button.

Answering calls with the handset. Pick up the handset and select a line using the left column or press Next to select the line which has been ringing longest.

Answering calls using the speakerphone. Press the **Speakerphone** button and select a line using the left column or press **Next** to select the line which has been ringing longest.

4.2 The Call Controller

The Call Controller, together with a user-provided telephone, emulates a Desktop Director. See the Call Controller manual for additional information.

Because there is no LCD, the dialing keypad is used to change modes, as follows:

*PRODUCER (*77638237) changes to Producer mode

*TALENT (*825368) changes to Talent mode

*DD (*33) is current ID for this unit

*SHOW1 (*74691) connects this controller to Show 1

*SHOW2 (*74692) connects this controller to Show 2

4.3 The Console Director

Telos provides sub-assemblies to a number of mixing console manufacturers, allowing them to offer you a console-mounted control surface compatible with the Nx12, 2x12, and Series 2101 systems. The Console Director works much like the Desktop Director in Talent mode.

5 Detailed Configuration & Reference

First Step...

After power-up and initialization, the status screen appears.

When you see this screen, and if

- ISDN or POTS lines are connected and functioning correctly, and
- the Nx12's basic configuration settings have been adjusted to the requirements of your installation,

then the Nx12 is ready for next steps.

You may confirm that ISDN lines are ready with the ISDN status screens. Press any menu button twice and use the up and down arrows to scroll through these screens. If all is OK, you will see the Ready indication for each ISDN B channel.

If you do not see READY as the ISDN status, first cycle power to the Nx12 off and then back on. If you still do not seen READY on the ISDN status screen, review the information in Sections 2.4, 2.5, and 3.2 and check both your ISDN settings and set-up information. If the information there does not help, there is in-depth troubleshooting information in Section 6.

Tip

Users in the USA & Canada should see READY status whenever the system is idle. On certain ETS-300 ISDN lines the system will go from READY to DEACTIVATED. This is no cause for alarm.

5.1 Level Metering

The Nx12 includes four LED bargraph meters. The ballistic (time) characteristic of these meters changes depending on the Audio menu Input Level setting to match the expected console meter type. In VU mode, they have the usual 300ms time-constant. In PPM mode, they have a 10ms attack and 2.8sec release characteristic.

The meters are selectable to various points in the system with Meter Mode in the Audio menu. Note that the marked meter names are correct only when the **Studio 1&2** mode is selected. See Section 5.4 below.

5.2 Status Displays

Telco line and Desktop Director status information is available on both the Web interface and the LCD.

On the LCD, pressing any menu button twice displays the status screens. These status screens display information about the state of the Nx12, but do not allow you to make any changes. The Δ & ∇ buttons cycle through the following screens:

- Slot A through B: Line Status
- Desktop Director Connection Status 1-8

Line Status

This shows the status of the Telco channels, helping you to confirm that the Nx12 is successfully configured for each. In addition, information about call status is indicated. One of the following words will be shown for each Telco channel present:

Deactivated
No line connected, or connected to wrong jack on the Nx12. Or ETS 300 permits the Telco to put a line to sleep. Deactivated will be displayed when this occurs. Or system is currently testing for the presence of a POTS line (e.g. looking for loop current)

SPID Pend
SPID has been sent to network (DMS Cust. And Natl I-1 only). Network has not approved SPID (SPID is incorrect).

SPID Error
The SPID has been sent and rejected by the CO switch.

Ready
The unit sees a live ISDN S interface (i.e. external NT1 is present or internal NT1 has completed startup). Or loop current has been detected on a POTS line and the line has been declared 'in service' by the system.

Pending Seize
Outgoing call in process; Request for call has been sent to Telco switch. Or outgoing POTS call is pending – system if 'off-hook' and waiting for loop current.

Seized
Outgoing call in process; Telco switch has acknowledged the call request. Or Loop current has been detected on a POTS line after going 'off-hook'.

Call Proceeding
Outgoing call in process; Telco switch waiting for routing info from network.

Outgoing Ring
Outgoing call in process; far end has generated a call alert signal

Disconnecting
Request to disconnect has been made. Awaiting acknowledgment. Or the POTS interface has gone 'on-hook' and system is waiting for loop current to cease.

Incoming Ring
An incoming call request has been detected and is being processed.

Accepting Call
Nx12 has accepted an incoming call and is waiting for the Telco switch to connect it. Or incoming POTS call is pending – system is 'off-hook' waiting for loop current.

Connected
All signaling is complete – the ISDN voice channel is available for use. Or a POTS line is 'off-hook' and loop current has been detected.

Desktop Director Status

This screen shows the status each Desktop Director. CH represents the channel number. As it is possible to connect two Desktop Directors on the same port, there are two channels per port. These are numbered left to right as viewed form the rear of the unit, with two channels per port.

Each channel is assigned a Desktop Director number: DD notification. Thus, by looking at the Director number on its LCD (upper left corner), it possible to find out exactly which port this one is connected to. One of two messages will be displayed:

Activated
The Desktop Director is up and running on this channel.

Deactivated
No Desktop Director is connected to this channel, or if it is, it is not being recognized by the Nx12.

Whenever a Desktop Director is activated, an Errors number is indicated. This corresponds to the number of data packets lost since this Desktop Director was powered on. An occasional error is not cause for alarm, however if you see multiple errors accumulating over time, this should be investigated. Possible causes are bad wiring, a bad port on the Nx12, or a defective Desktop Director.

> **Note**
>
> **If a Desktop Director Is connected to an external power supply, unplugging the connection at the back of the Nx12 will not reset the error counter. A few errors (2-3) when powering up the Nx12 is normal. If a Desktop Director Is 'frozen', the Nx12 will deactivate it. If this ever happens, unplug the Desktop Director, then plug it back in.**

On the Nx12 LCD, it is possible to investigate further by pressing the OK button. This will give access to a detailed Diagnostics menu for each Director.

Select any available Desktop Director connection: DD connection

The Far end menu gives a detailed error status coming from the Desktop Director side. The Near end menu gives a detailed error status coming from the Nx12 side.

> **Tip**
>
> **If you suspect a malfunctioning Desktop Director, check its error status before calling Telos support. Having these details at hand will allow us to best assist you.**

5.3 Using the LCD/Buttons and Web for Configuration

You use the various configuration settings to adjust audio levels and processing options, set the ISDN parameters, etc.

You may use either the front panel LCD/button user interface on the Nx12 main unit or the Web interface to access configuration options. The Web interface offers the full set of configuration options, while the LCD/button includes only the most often needed items.

The LCD/Button User Interface

Each of the three buttons below the LCD display calls up the menu item group associated with the button's name.

Pressing any menu button (**Audio**, **Telco**, or **System**) twice displays the Status screen.

Menu Navigation & Item Selection

After you select a menu group, you can use the Δ and ∇ arrow buttons to scroll through the available items. You may select a highlighted item for modification by pressing the OK button. Once an item is selected, you can use the Δ and ∇ arrow buttons to can change the value of the setting.

When you reach the value you desire, press the OK button again to lock it in and return you to the item selection/scroll mode.

The Web User Interface

A Web Browser can be used to remotely configure and administer the system. All front panel configuration options are duplicated on the web pages. Certain advanced options such as entering ISDN DNs/SPIDs and Livewire configuration require the Web interface.

You can connect a PC directly to the Nx12 with an Ethernet cross-cable. Or you can connect both the PC and the Nx12 to an Ethernet switch or hub.

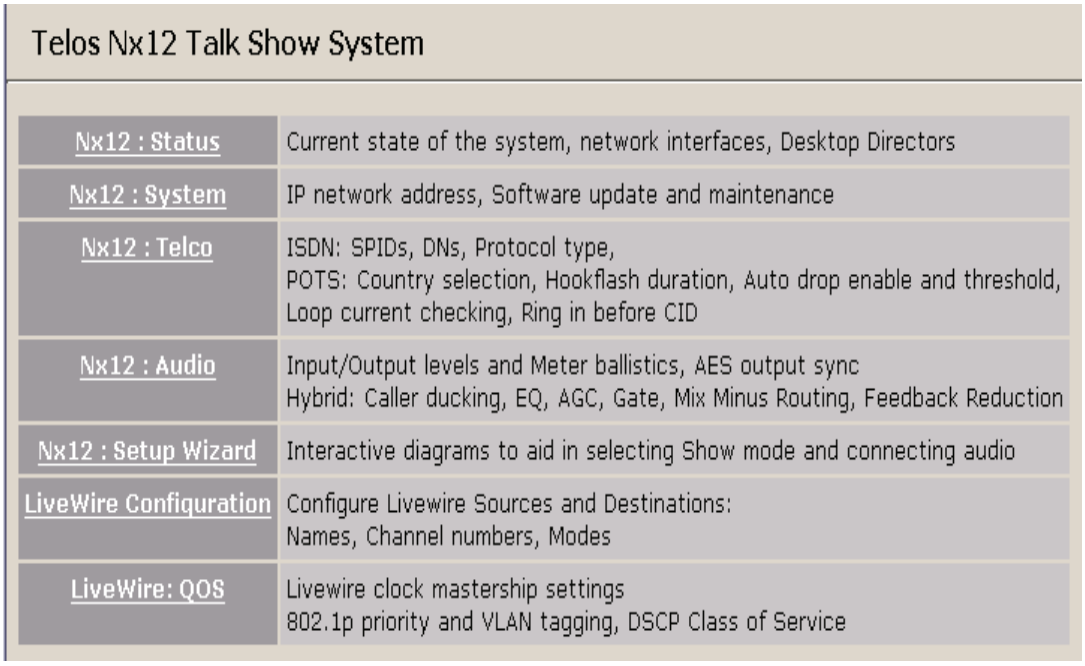
Important Note

If Livewire has been enabled, there will be a lot of network traffic generated by the Nx12. Some PCs, switches, hubs, and routers that are not multicast-aware will not be able to handle the volume and Web communication will not be possible.

Normally, you would use the Web interface to switch off the Livewire streams, but since you can't communicate with the Web interface, you will have to first disable the Livewire streams from the front panel LCD/button interface. The first item under the Audio menu is what you need.

The Nx12's default IP number is **192.168.0.230**. Normally, you would change this to a number appropriate for your network using **TCP/IP Setup** in the System menu. You would also enter appropriate values for Subnet Mask and Gateway Address. See The System Menu below for details. But if you will just be doing configuration with a cross-cable or isolated switch/hub, you can keep the default IP number. You can also connect first with the default, then change the IP values using the Web, then reconnect with the new address.

After you have a network connection between your PC and the Nx12, open a Web browser and enter **http://192.168.0.230** (substitute your actual IP number if you have changed from the default) into the address entry field. You will then see the Nx12's main Web screen:


Telos Nx12 Talk Show System
Nx12 : Status
Current state of the system, network interfaces, Desktop Directors
Nx12 : System
IP network address, Software update and maintenance
Nx12 : Telco
ISDN: SPIDs, DNs, Protocol type,
POTS: Country selection, Hookflash duration, Auto drop enable and threshold,
Loop current checking, Ring in before CID
Nx12 : Audio
Input/Output levels and Meter ballistics, AES output sync
Hybrid: Caller ducking, EQ, AGC, Gate, Mix Minus Routing, Feedback Reduction
Nx12 : Setup Wizard
Interactive diagrams to aid in selecting Show mode and connecting audio
LiveWire Configuration
Configure Livewire Sources and Destinations:
Names, Channel numbers, Modes
LiveWire: QOS
Livewire clock mastership settings
802.1p priority and VLAN tagging, DSCP Class of Service

Upon clicking any of the menu links, you will be prompted to enter a user name and password. The default user name is **user** and the password is <blank>.

The Status, System, Telco, and Audio menus provide access to the full set of configuration options, a superset of those available on the front panel. The ISDN SPID and DN entries are only available on the Web. interface (It's not very convenient to enter numeric data on the LCD/button interface, so we thought you'd appreciate having a PC's keyboard for that.)

The Livewire menus are only available on the Web interface.

5.4 The Audio Menu

This group consists of settings that control the audio levels, processing options, mixing, etc.

Nominal Input Level

Adjusts to the level expected from the mixing console. This adjustment affects all of the input options: analog, AES, and Livewire. It is applied before the send AGC.

Be careful to avoid clipping of the Nx12 inputs. Clipped send audio will create a problem for the system, and you will probably experience poor hybrid performance. Increasing the input level past the correct value will not make the send louder due to the AGC/limiter.

Input Setting Related to Input Level

Input Setting	-10dBV	0dBu	+4dBu	+9dBu PPM	-9dBfs PPM
Analog input level	- 10 dBV nom	0 dBu nom	+4 dBu nom	+9 dBu PML	n/a
Livewire/AES	- 32 dBfs nom	-24 dBfs nom	-20 dBfs nom	-15 dBfs	-9 dBfs PML

The ballistic (time) characteristic of Nx12's meters changes depending on the Input Level setting so to match the expected console meter type. When -10, 0, or +4dBu is selected, the meter has a VU-standard integration time. When +9dBu PPM or -9dBfs PPM are selected, the meter has a PPM time characteristic.

The table above shows the level at the green/red transition on the NX12 meter. For VU, this is the nominal 0 value; for PPM, this is the Permitted Maximum Level (PML) value.

The usual setting is +4dBu. The +9dBu PPM setting is intended for Euro analog installations. The -9dBfs setting is intended for Euro digital installations.

If Livewire inputs are used, this should usually be set to +4dBu for the -20dBfs Livewire standard nominal level. Europeans could choose the -9dBfs PPM setting if they are operating their Livewire network according to this standard.

Nominal Output Level

This adjusts the level of analog, AES, and Livewire outputs.

Output Setting Related to Output Level

Output Gain Setting	-10dBV	0dBu	+4dBu	+9dBu PPM	-9dBfs PPM
Analog output level	- 10 dBV nom	0 dBu nom	+4 dBu nom	+9 dBu PML	n/a
Livewire/AES	- 32 dBfs nom	-24 dBfs nom	-20 dBfs nom	-15 dBfs	-9 dBfs PML

Meter Mode

Selects the audio that will be indicated on the LED meters.

- **4 Studio Out** – Each studio output goes to a meter, from the top down
- **4 Studio In –** Each studio input has a meter, from the top down
- **POH in –** The two Program on Hold inputs go to the top meters
- **Phone 3&4 –** Send and receive to/from hybrids 3 & 4
- **Phone 1&2 –** Send and receive to/from hybrids 1 & 2
- **Studio 3&4** – Input from and output to studios 3 & 4
- **Studio 1&2 (default)** – Input from and output to studios 1 & 2 (the meter markings correspond to this one)

Phone Send Adjustment

This setting adjusts the level of the hybrid into the telephone line, after the send limiter and is calibrated with 0dB representing the usual send level into the line.

The default is 0dB, which should be OK for most telephone lines. If callers consistently complain about low audio level and you have verified that this is a problem, you can increase the

send level. However, doing so can reduce effective hybrid performance and increase leakage. Conversely, lowering this setting can reduce leakage.

Call Ducking

This control adjusts the amount of ducking that occurs to the received telephone audio when send-to-caller audio is present. A value of **Full Duplex** means that the telephone audio is never ducked or attenuated, while **Half Duplex** applies maximum attenuation when send audio is present. Intermediate settings range from -1 to -15dB. Section 5.11 has more information on this function. Ducking is desirable for a variety of reasons:

- Allows the announcer to 'override' the caller by causing the received caller audio to be attenuated (ducked) when the announcer speaks. This is often desired for aesthetic effect and allows the announcer to remain in control of the conversation.
- Helps to have open loudspeaker monitoring of callers without feedback.
- Dynamically improves effective trans-hybrid loss to reduce leakage, when necessary.

There is also a ducking of the announcer audio to the caller. This is only a few dB unless Feedback Reduction is enabled, as described below. This send ducker has a fixed value.

Receive AGC

The receive Automatic Gain Control serves to improve level consistency of caller audio. Nominal levels of the telephone network vary as much as 30dB from call to call. In the past decade our experience has been that loud calls have become just as much a problem as low level callers. Therefore we have changed the AGC to act as a platform leveler. The Nx12 uses a dB-linear approach to AGC with a feed-forward topology, which provides a consistent sound regardless of the drive level. This "smart compressor" normalizes levels while retaining the natural dynamics of the caller's voice.

This selection controls the "aggressiveness" of the AGC process, taking values from **OFF**, **1/4**, **1/2**, **3/4** to **FULL**. We suggest you start with the default value, **FULL**

Aggressive AGC is more noticeable, but is also more effective at maintaining consistent levels. Adjustment of this setting simultaneously changes as number of parameters within the AGC function, such as attack & release times, thresholds, and compression ratio.

An important feature of the Nx12's AGC is that it is cross-coupled to other sections of the hybrid so that it can reliably distinguish between the real caller audio and residual hybrid leakage. This allows a lot of gain on low level callers while preserving excellent hybrid performance.

Noise Gate

Enables or disables a noise gate applied to the caller audio. This is a "downward expander" that has the effect of reducing the noise on the caller audio when it falls below a threshold. On some very low-level calls, this attenuation may be inappropriate because you may be able to hear the caller fade away when the level falls below the threshold. It has three settings as follows:

- **Off** – Disables the noise gate
- **Normal** – A slow setting, with moderate noise attenuation
- **Aggressive** – A faster setting with deeper attenuation, for lines where noise is noticeable during normal conversation.

Receive EQ (Caller Dynamic Digital EQ – DDEQ)

Telephone audio frequency response varies widely since many factors can affect it (we've measured the response on a number of calls and the results were revealing). Consequently, some form of receive equalization is desirable. There are three choices for the type of equalization applied to the received telephone audio:

- **Off** – The caller audio is passed through without modification. The Rcv EQ values are ignored in this mode.
- **Fixed** – This is a simple manual equalizer mode where a fixed EQ is applied.
- **Adaptive** – This is a three band dynamic equalizer. The high and low frequency boost or cut is automatically determined and set. The Dynamic EQ uses the values set below as 'target levels'.

(Additional) Low EQ

Amount of low frequency boost or cut applied to the caller audio. Settings are in 2dB steps from -8dB to +8dB are possible with 0dB representing flat response. The default value is +4dB, which sounds good across a variety of callers, lines, and telephone sets.

(Additional) High EQ High

Amount of high frequency boost or cut applied to the caller audio. Settings from -8dB to +8dB are possible with 0dB representing flat response. The default value is 0dB.

Feedback Reduction

This enables or disables the acoustic adaption process used for feedback control when the system is used with an open speaker-to-microphone acoustic path. This setting also increases by 6dB the amount of send to caller audio ducking. It should be set to On when it is needed to eliminate feedback. Since the send to caller audio will be colored by this process, should be turned off unless needed. Default is Off.

When using open speakers it is best not to change the speaker or mic positioning while the system is operating as this will require the acoustic canceller to re-adapt and temporary feedback may occur.

Mix Minus Mode

*On the Web interface, the **Setup Wizard** can be used to set the Mix-Minus routing mode.*

This option determines how many mix-minus feeds you will need (see section 2.7 for more on mix-minus) and the number of fader channels required on the mixing console.

The effect of the mix-minus mode depends upon the Show mode, which can be One Show, 4-Column, or one of the Two Show modes. See section 5.7 for more about Show modes.

The mix-minus mode options are:

- **Separate (default)**

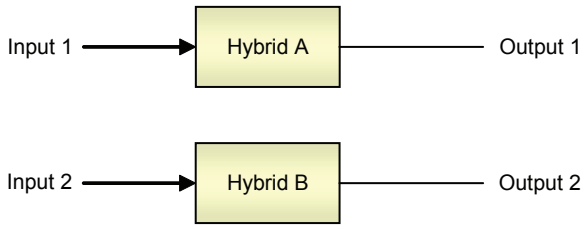
  This option provides fully independent inputs and outputs for each hybrid. Thus, you will feed the number of faders on your console equal to the number of hybrids your show has, and your console will have to create the same number of mix-minus feeds back to the Nx12.

  In this mode, callers cannot hear each other except through an external mix-minus. Thus,

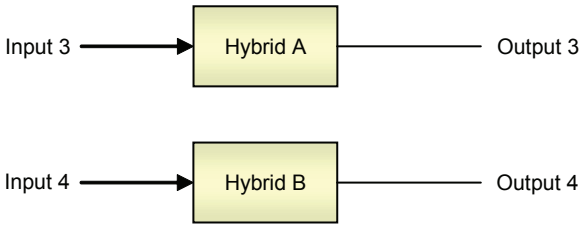
Separate requires the mixing console to provide the cross-connection for conferencing, when required.

With One Show show mode, and in any of the Two Show show modes, there are two hybrids per show, so using the Separate mix-minus mode means you have two caller outputs to each console, and two mix-minus caller inputs from each console.

One Show, or 1st of two Shows:



Second of two Shows:



In 4-Column show mode togther with Separate mix-minus mode you will have four independent caller outputs to four faders on the console, and you will need four mix-minus feeds.

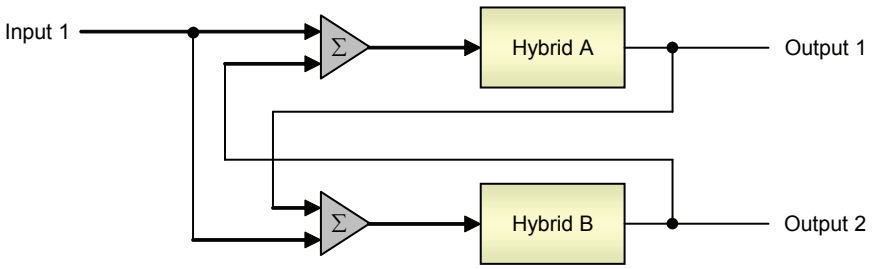
- **1 in/multiple out**

  Using this option means each mixing console only has to make a single mix-minus feed for the caller send audio for each show, but separate hybrid caller audio outputs are sent to the console. This mode is useful when your console is limited in the number of mix minus feeds it can make, but you want to keep individual faders per hybrid.

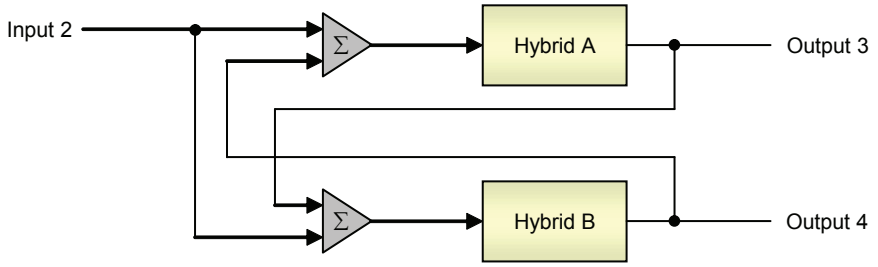
  This option sends the single mix minus feed from the console to each caller. Internally within the Nx12, the cross-connected audio from the other callers is mixed in, so each caller can hear the other callers.

  With One Show show mode, and in any of the Two Show show modes, there are two hybrids per show, so using this mix minus mode means you have two caller outputs to each console, and one mix minus caller send from each console.

  One Show, or 1st of two Shows:



Second of two Shows:



In 4-Column show mode, you have 4 independent caller outputs to 4 faders on the console, but require only 1 mix minus feed. All callers hear the one console mix-minus feed and each other.

- **1 in/1 out**

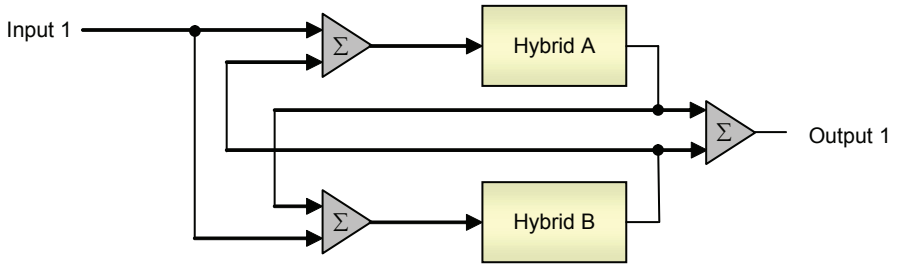
This mix-minus option means each console only has to make a single mix-minus feed for the caller send audio and the Nx12 makes only one mixed caller audio output of all callers, to one fader on the console.

Like the 1 in/multiple out mode, this option sends the single mix-minus feed from the console to all callers.

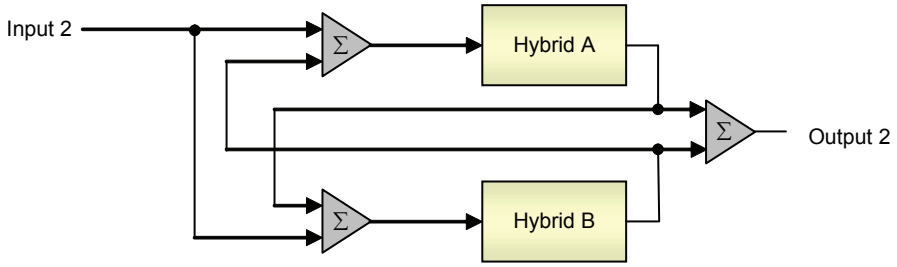
Its main purpose is to accommodate older consoles that have only one mix-minus, or when only one fader is available for telephone calls. The disadvantage is that the operator has no control of the relative levels of the multiple callers. It relies upon the Nx12's AGC to automatically adjust the multiple callers to a consistent level.

With One Show show mode and in any of the Two Show show modes, using this mix-minus mode means you have one caller output to each console, and one mix-minus caller input from each console.

One Show, or 1st of two Shows:



Second of two Shows:



With 4-Column Show mode, you will have one caller output to the console containing a

combined mix of all callers' audio, and require only one mix-minus feed from the console. All callers hear the one console mix-minus feed and each other.

The following table summarizes the effect of each mix-minus mode together with each Show mode:

	Mix-Minus Mode *Separate*	**Mix-Minus Mode** *1 in/Multiple out*	**Mix-Minus Mode** *1 in/1 out*
Show Mode *One-show*	**2 in, 2 out, 1 POH** **Use analog, AES, or LW.** **For AES, POH may be analog or AES.**	**1 in, 2 out, 1 POH** **Use analog, AES, or LW.** **For AES, POH may be analog or AES.**	**1 in, 1 out, 1 POH** **Use analog, AES, or LW.** **For AES, POH may be analog or AES.**
Show Mode *One-show, 4-column*	**4 in, 4 out, 1 POH** **Use AES in 4-channel mode or LW to access all the inputs and outputs.** **For AES, POH remains analog.**	**1 in, 4 out, 1 POH** **Use AES in 4-channel mode or LW to access all the outputs.** **With AES in 4-channel, POH remains analog.**	**1 in, 1 out, 1 POH** **Use analog, AES, or LW.** **For AES, POH may be analog or AES.**
Show Mode *Two-show*	**Per show: 2 in, 2 out, 1 POH** **Use AES in 4-channel mode or LW to access all the outputs.** **For AES, POH remains analog.**	**Per Show: 1 in, 2 out, 1 POH** **Use AES in 4-channel mode or LW to access all the outputs.** **For AES, POH remains analog.**	**Per show: 1 in, 1 out, 1 POH** **Use analog, AES, or LW.** **For AES, POH may be analog or AES.**

Livewire can always be used for any of the four caller and two POH inputs. When a Livewire input is enabled, the corresponding analog/AES input is disabled. Livewire can provide inputs 3 and 4 when they are not otherwise available at the XLR connectors.

Four caller audio outputs via Livewire are always possible, regardless of mode, in parallel and at the same time as the XLR audio outputs.

AES Mode

*On the Web interface, the **Setup Wizard** can be used to set the AES mode.*

If your Nx12 is equipped with the optional AES module, there are two AES input and two output XLRs, which may be configured as follows:

- **2-channel (default)**

  In this mode there are up to two caller inputs/outputs.

  The two AES input XLRs can either take one audio input each using only the L channels, or a combined L/R AES signal may be connected to the first AES input XLR.

  The Nx12 will automatically sense if there is a second AES connection to XLR input 2. If present, the Nx12 will take the second audio input from this XLR. If not present, the Nx12 will take the second audio input from the R channel of AES input 1.

  If your mixing console combines two caller sends into one AES L/R, plug this into the first AES input XLR and use the first AES output XLR for the caller audio. If your console uses two separate AES connections, use both XLR inputs and outputs.

  Program on Hold is via the analog inputs.

- **4-channel**

  In this mode there are up to four caller inputs/outputs.

  The L and R of each AES signal are used, creating four independent caller inputs and four caller outputs.

  To use this mode, your mixing console must be capable of splitting and combining separate channels of audio to and from the L/R pairs of AES signals. This is sometimes called the 'dual mono' mode of AES operation.

  Program on Hold is via the analog inputs.

- **AES POH (AES Program On Hold)**

  In this mode there are two caller inputs/outputs and two AES Program on Hold inputs.

  In the case when only two caller inputs/outputs are needed and the mixing console is capable of splitting and combining separate channels of caller audio to and from the L/R pairs of AES signals, this mode allows the Program on Hold to also be via AES. The caller inputs use AES XLR Caller input 1 (both L / R channels), and Program on Hold inputs use AES XLR input 2 (both L / R channels).

The following table shows the relationships of AES to the input and output signal paths shown in the block diagrams above:

	2-channel (1 XLR connection)	*2-channel* (2 XLR connections)	*4-channel*	*AES POH*
AES in 1 L	"input 1"	"Input 1"	"Input 1"	"Input 1"
AES in 1 R	"Input 2"		"Input 2"	"Input 2"
AES in 2 L		"Input 2"	"Input 3"	POH in 1
AES in 2 R			"Input 4"	POH in 2
AES out 1 L	"Output 1"	"Output 1"	"Output 1"	"Output 1"
AES out 1 R	"Output 2"	"Output 2"	"Output 2"	"Output 2"
AES out 2 L	"Output 2"	"Output 2"	"Output 3"	"Output 2"
AES out 2 R	"Output 1"	"Output 1"	"Output 4"	"Output 1"

5.5 The Telco Menu

POTS Country

Affects analog lines only.

This setting configures the analog Telco line interfaces for proper impedance matching and other characteristics for different countries. If your country is not listed, try using: CTR21 (European standard) or USA.

Hookflash

Affects analog lines only.

It adjusts the duration of the 'hook flash' (loop current interruption) when the Desktop Director (transfer) key is pressed. The default value is 720 milliseconds. The following table indicates typical values for various countries:

USA	**720 ms**
Belgium, Netherlands, Switzerland	**120 ms**
France, Luxemburg, Portugal, Germany	**270 ms**
Rest of Europe	**90 ms**

The hook flash is sometimes used as a signal to initiate transfer or other features, with the specific effect being dependent on the Telco or PBX capabilities and configuration.

Loop Current Checking (Web only)

Affects analog lines only.

The Nx12 periodically checks each analog line port for loop current. When current is detected, the single green dot icon appears on Desktop Directors and the line is ready for use. This process is called Loop Check. Yes is the default.

When you are using 'ground start' lines, this should be set to Off. Note: The Nx12 does not support outbound dialing on ground start lines. However, setting Loop Check to No will allow incoming calls to work normally.

Note

Ground Start analog lines are not fully supported by the Nx12. Only Incoming calls are allowed. You can take advantage of this to get a feature that might be sometimes useful: preventing outgoing calls.

Enable Auto Call Drop (Web only)

Calls can be automatically disconnected when the far end hangs-up. Most USA-style Telco lines provide a momentary loop drop for this purpose, but non-USA lines and most PBXs provide a tone signal (Call Progress Tone - Disconnect) instead. The Nx12 can respond to one, both, or neither of these signals. The type of tone signal detected for CPTD is determined by the POTS Country setting.

CPC loop drop threshold (ms) (Web only)

Sets the threshold time value for loop drop disconnect.

Ring-in Before Caller ID (Web only)

Normally the ringing indication is passed to the Desktop Director and other devices after the Caller ID has been received. This delays the ring indication until after the 2nd ring. You may want to have a faster ring indication, and choosing yes here lets you have it. But you risk losing the CID if an operator answers the call quickly. Some lines don't have CID and in this case this should be set to Yes since there is no reason to wait for the 2nd ring.

ISDN Protocol

Selects the ISDN protocol, which must correspond to the ISDN circuits you are using. It will apply to all ISDN lines.

There are a number of choices, but the most common are described below.

For Europe and many other parts of the world

- ETS 300. This is the very common Euro-ISDN, according to the ETS 300 standard.

For USA & Canada and a few other parts of the world

- Natl. I-1. National ISDN-1 is the most common protocol for the USA & Canada and some other parts of the world. May also be used with lines conforming to National ISDN-2. See the notes about SPIDs, below.
- DMS Cust. For 'custom functional' lines on the Nortel DMS-100 switch. See the notes about SPIDs, below.
- AT&T Cust. This is the AT&T PTP protocol, sometimes called 'AT&T custom Point-to-Point'. This protocol generally does not allow two simultaneous voice calls. For this reason we recommend that the National ISDN-1 protocol be used if you are on the AT&T/Lucent 5ESS Telco switch.

If you are in Europe or most of South America, you should choose ETS 300.

If you have Net64 ISDN in Japan you should select INS64.

If you are in Hong Kong you should choose AT&T Cust.

If you are in the USA or Canada, use whichever protocol your phone company has provided. If you do not know, you may be able to guess...

Guessing the ISDN Protocol setting in Canada & the USA

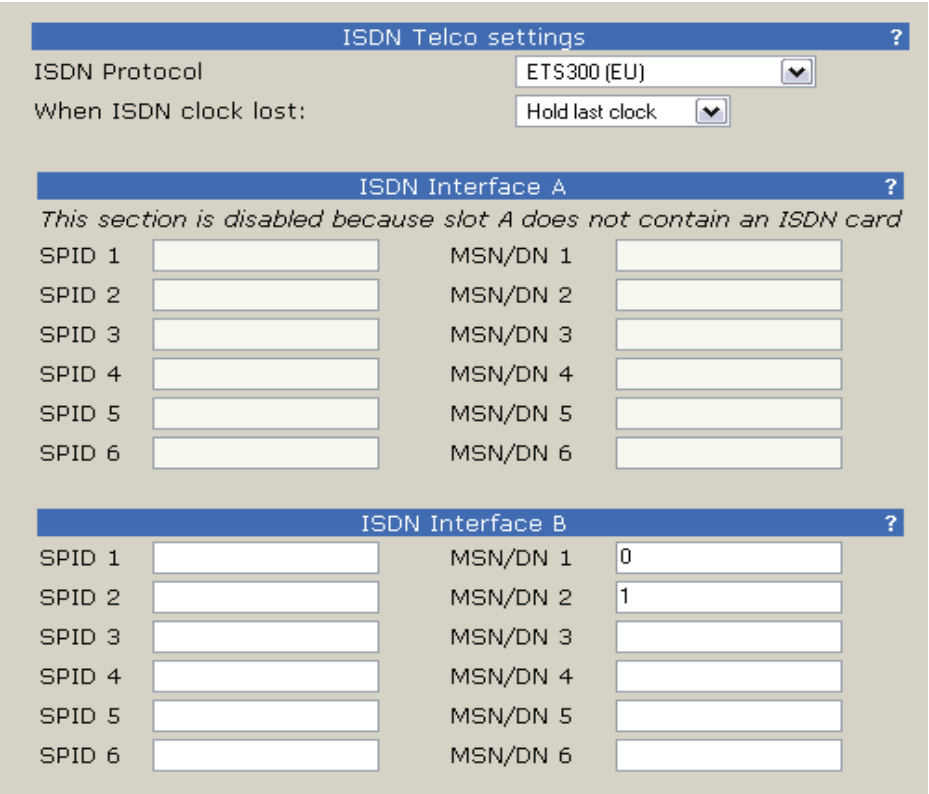
- If you were given only a single phone number and no SPIDs try AT&T Cust
- If you were given 2 SPIDs with the format: 01+7 digit phone #+0 try AT&T Cust
- If you were given 2 SPIDs with the format: 01+7 digit phone #+000 or 01+7 digits +011 try Natl.I-1
- If you were given 2 SPIDs with the format: area code+7 digit phone #+4 digits (for example 0101, 0100, 0111, or 1111) try Natl.I-1. If that fails Try DMS Cust
- If you were given 2 SPIDs with the format: area code+7 digit phone #+1 digit (for example 1, or 0) try DMS Cust. If this fails try Natl.I-1
- If you were given 2 SPIDs with the format: area code+7digit phone #+2 digits (for example 01, 11, 02, or 00) try DMS Cust. If that fails try Natl.I-1
- If you were given 2 SPIDs with the format: area code+7digit phone #+3 digits (for example 000,001) try Natl.I-1. If that fails try DMS-Cust
- If you were given a single phone number and a single SPID with the format: of 01+7 digit phone#+0 try AT&T Cust (Do not enter this SPID)
- If you were given a single phone number and a single SPID with the format: of 01+7 digit phone#+00 try Natl.I-1. If that fails try AT&T Cust
- If you were given a single phone number and a single SPID with the format: of area code+7digit phone#+4digits (for example 0101, 0100, 0111, or 1111) Try Natl.I-1. If that fails try DMS

Note

After you enter (or change) your SPIDs or DNs, you must reboot the Nx12. Therefore, once you have entered SPID1, SPID2, DN1, and DN2 you should turn the power off and then back on. Then go to the Status screen and verify Ready status for both channels.

SPIDs and MSN/DNs (Web Only)

There will be entry locations for SPIDs and MSN/DNs for all line positions, whether needed, or not. After entering the numbers, you must click **Apply** and then **Reboot Now** at the bottom of the page to activate the ISDN lines.



(You can check what you actually have by looking at the first line of the LCD status screen, where you will see POTS, ISDN, or EMPTY for the A and B line groups, depending on how your system is configured. Press any menu button twice to get this screen.)

When you are using Natl. I-1 or DMS-100 for your ISDN Protocol mode, you must enter the correct Service Profile ID (SPID) numbers for each ISDN channel, normally two per physical line. The telephone company should tell you this number when they confirm your order. Often it is found written on the line jack or punch block.

Usually the SPID is the area code+phone number+0101. However, we have seen some that include a two-digit prefix and a two-digit suffix, and many other variations are possible. See Appendix 1 for a list of known working SPIDs. If this works, consider yourself lucky; if it doesn't work, there is no substitute for getting the correct SPID from your Telco!

ISDN Tip

ETS 300 users should disregard all references to SPIDs. Euro ISDN does not have SPIDs (you lucky souls)!

IMPORTANT

If your line status shows "Wait" at least one of your SPIDs is incorrect or your ISDN Protocol setting is incorrect! In this case, check the SPID. For a list of known working SPIDs by telephone company, see the Appendix.

Notes on SPIDs

1. Incorrectly entered SPIDs are the most common problem USA users have placing ISDN calls. Compare the SPIDs you have entered with the SPIDs provided to you by the phone company to be certain that you have entered them correctly. Do not add anything to your SPIDs.
2. Note that there are no hyphens, dashes (-), or spaces in SPIDs, so the system will not allow you to enter any non-numeric characters. If your installer has included them in the SPID, ignore them.
3. If you have any questions about your SPIDs, call your phone company.
4. If you have AT&T Point-to-Point or ETS300 Euro-ISDN Protocol mode, you should not enter anything in the SPID fields.

MSN – Multiple Subscriber Number

This item, if required, must be configured from the Web interface. Used only when you are using the ETS 300 Euro-ISDN Telco mode and another ISDN device is sharing the line with the Nx12. MSN1 corresponds to the first B channel. This is simply one of your assigned telephone numbers. Normally, for ETS 300 these should be left blank.

DN – Directory Number.

This item, if required, must be configured from the Web interface. When you are using an ISDN protocol other than ETS 300 or AT&T Cust. PTP, you must enter the 'Directory Number' (DN) here. This is simply the telephone number associated with the channel.

Tip

Each directory number will correspond to one SPID. You must enter them to reflect this relationship. The DN corresponding to SPID1 must be entered as DN1 while the DN corresponding to SPID2 must be entered as DN2.

If you experience problems only with incoming calls check your DNs. Normally 7 digits should be entered, but you might need to enter 10 digits.

In some cases lines in a hunt group will require that the same (primary Listed Directory Number) be entered for all lines that are part of the hunt group.

Hot Tip

Some versions of the Telos Zephyr and Zephyr Express do not require that Directory Numbers be entered. This is not true for the Telos Nx12 and other newer products.. You must correctly enter these numbers for the system to work.

Note for the Curious

The actual physical B channels of the BRI line are assigned on a per call basis, with the first incoming call assigned to the first B channel. However, the call setup message on the D channel includes what number was dialed, and the Directory Number allows the system to sort it all out.

IMPORTANT

After you enter (or change) your SPIDs or DN's you must restart the system to use the new information. Therefore, once you have entered SPIDs, DNs, or MSNs cycle power off and then back on. Then go to the ISDN status screen and verify "Ready" status for all channels.

Clock Loss

This setting is relevant only when there are ISDN lines that "go to sleep" or there is some other unusual ISDN Telco condition. Most users should leave this set to the default setting of **Hold Last.**

5.6 The System Menu

This menu consists of items that affect basic system operation.

TCP/IP Setup

The Nx12 and PC hosting the Web browser should have IP addresses in the same range (the first 3 of the four numbers of the IP address should match) and should have the same subnet mask.

Host Name (Web only)
The IP network text name. Default is **Nx12.**

IP Address
This is the IP address for this Nx12. An IP address is a series of 4 numbers separated by periods. This is a numeric-only field; enter the digits as described above. You should enter no leading zero in each of the four numbers (if you are given 192.168.000.012 enter 192.168.0.12).

Subnet
This is your subnet mask. The subnet mask is set to the size of your local network. All packets addressed to a destination outside this local area are sent to the gateway router.

The subnet mask is a series of 4 numbers separated by periods in the same format as the IP address. A subnet with 256 addresses would have this mask value: 255.255.255.0.

Gateway
This is the IP address of a gateway router connecting your LAN to the Internet. You will need

to get this from your network administrator or Internet service provider.

To enter IP addresses using the LCD/button interface:
Select the **TCP/IP Settings…** sub-menu.

To enter an IP address, use the following procedure:

1. Highlight the IP address
2. Press OK
3. Press ∇ repeatedly to delete existing IP address
4. Press Δ until the desired digit is displayed
5. Press OK to accept that digit. To delete a incorrect digit press ∇
6. Press Δ until the desired digit is displayed
7. **Once all characters are entered, press OK twice to save your entry**

Return takes you back to the System menu.

Syslog Server & Syslog Filter (Web only)
Permit logging to external computers running Linux standard Syslog for troubleshooting purposes.

Contrast (LCD Only)

Adjusts the contrast of the LCD. Values range from 0 to 10. The usual setting is 5.

Show Mode (LCD Only)

*The Web **Setup Wizard** is another way to select show mode.*

The term "Show" comes from our Series 2101 system, which supports up to 120 lines and more than 64 hybrids in multiple studios. Each Show selects a group of Telco lines and an audio I/O. Next queues and other functions are distinct for each show.

The Nx12 has only 12 lines and 4 hybrids, so it's considerably simpler. Single studio modes have only **Show 1**. However, the Nx12 also supports two-studio modes, in which case there will be **Show 1** and **Show 2**. Desktop Directors and call screening applications connect to the appropriate show, and each show has associated audio inputs and outputs.

See Section 5.7 for details on the various show modes. Choices are:

- One Show (default)
- 2 Show Custom
- 2 Show Split
- 2 Show Shared
- 4 Column

Note: You must reboot the system after changing this setting.

Hold on Select

The Direct Hold feature allows the operator to seize the next call which has been ringing for the longest time and put the caller he was talking with back on hold in one simple step, by pressing the **Next** key.

If no call is ringing, the process remains the same with callers on hold. Pressing the **Next** key seizes the call, which was on hold for the longest time, and puts the previous caller on hold.

This feature also works when manually selecting a new line in a column. If existing unlocked calls are present, the lines will be placed on hold.

The choices are:

- **None** – feature disabled
- **Talent** – enabled only for Desktop Directors in Talent mode.
- **Producer** – enabled only for Desktop Directors in Producer mode.
- **Both** – enabled for both Producer and Talent modes.

Audio Loop (LCD Only)

This is for testing purposes. Choices are:

- **Off** – This is the normal operating mode. No loopback path is turned on. Studio Input is sent as send audio, and caller audio is presented at Studio Out.
- **Studio** – This loops the AES/EBU input (and analog signal after Analog-to-Digital conversion) back to the AES/EBU output (and analog output via the Digital-to-Analog converter). This can be used to proof the audio input and output sections
- **No EC** – This is not a loop mode, but a test mode. It disables the echo canceller on the phone line, so that measurements such as SNR and THD can be made from the Telco line to the output.

Factory Defaults (LCD Only)

Selecting this option displays the Restore Defaults sub-menu with the following options.

Return
This takes you back to the System menu.

Restore Defaults?
This menu has two options:

- **No** – Goes back to the Restore Defaults sub-menu.
- **Yes** – Erases all user configuration and sets all to the original factory defaults.

Reboot (LCD Only)

Selecting this item displays the Restore Defaults sub-menu with the following options.

Return
This takes you back to the System menu.

Reboot System?
This menu has two options:

- **No** – Goes back to the Reboot System sub-menu.
- **Yes** – Causes the system to reboot. (All callers will be lost.)

This command is intended to save you a trip to the back of the rack. This reboot is equivalent to removing power and then restoring it.

Firmware Update (Web Only)

See Section 5.9.

5.7 Show Modes

Recall that the options for Show Modes are:

- **One Show**
- **4 Column**
- **2 Show Split**
- **2 Show Shared**
- **2 Show Custom**

Note: You must reboot the system after configuring this setting. Be sure to attach the phone lines before rebooting.

Single-studio Modes

One Show (default)
Four hybrids are used to improve conferencing quality, but they are summed and mix-minused internally to appear as if they were two. (Or one, depending upon Mix Minus mode.) Each Desktop Director column can use up to two hybrids, with the first call on the left column connecting to Hybrid A, the second to Hybrid B, the third to Hybrid A, the fourth to Hybrid B, etc.

Only POH input 1 is used.

4 Column
This supports Desktop Directors with the add-on module to make four line-select columns.

Each column is connected to one of the four Nx12 hybrids, the leftmost to A, the next to B, etc. The four inputs and outputs are only accessible via AES and Livewire, and 4 Column mode cannot be used with the analog I/O option.

This also supports a Euro-style operation method where each of four Telco lines has its own hybrid and fader. As this is written, this mode is only possible with Euro-phone modules installed in an Axia Element mixing console.

Only POH input 1 is used.

Sharing the Nx12 – Two-studio Modes

These modes allow a single Nx12 to be used with two studios. Telephone lines may either divided and used independently or be shared with both studios having access to the same lines.

Each Desktop Director, Console Director, and call control software application needs to be attached to a show. See below for instructions on how to do this.

The POH 1 input goes to Show 1 and the POH 2 input to Show 2.

Note: When using the two-studio modes, you will need to use **Separate** for the Mix Minus selection in the Audio menu. See Section 5.4.

2 Show Split

This option causes the system to be split as follows:

- Show 1 (Hybrid A&B) has access to lines 1 through 6. Lines 1 through 5 are 'normal' lines while Line 6 is a 'priority' line. Line 6 is not affected by the 'Block All' function.
- Show 2 (Hybrid C&D) has access to lines 7 through 12. Lines 7 through 11 are 'normal' lines while Line 12 is a 'priority' line. Line 12 is not affected by the 'Block All' function.
- In any of the two-studio modes, the system maintains two separate Next and Screened Next queues. The two studios can be operated completely independently.

2 Show Shared

This option causes the system to be split as follows:

- Show 1 (Hybrid A&B) has access to lines 1 through 12. Lines 1 through 11 are 'normal' lines while Line 12 is a 'priority' line. Line 12 is not affected by the 'Block All' function.
- Show 2 (Hybrid C&D) also has access to lines 1 through 12.

When a line is shared between two studios, it becomes unavailable in one studio if it is in use in the other studio. In this case, the Status Symbol for this line will show a small 'x' on the other studio's Director. If a shared line is placed on hold, it can be picked up in either studio. Users of the Telos Series 2101 system will recognize this behavior, as that system works in the same way.

2 Show Custom

This lets you configure each line individually with regard to studio assignment and priority, thus dividing the lines between the studios according to your specific requirements. This programming is done from a Desktop Director attached to the system. If custom line configuration is performed via the Desktop Director, Show Mode will change to 2 Show Custom automatically.

To assign lines:

1. Press the MENU softkey on the Desktop Director.
2. Press <∇> repeatedly until **Configure Shows** is highlighted and then press <SELECT>.
3. Enter the password and then press the OK softkey. **Note**: The default password is **1234**.
4. Press <∇> to highlight Set Show Lines and then press <SELECT>.
5. Use <∇> and <Δ> to highlight each line. For each line Use the <+> or <-> Softkeys to select from the following options:
   - **NONE** – Line is not available for either show
   - **SHOW 1** – Line is available for the show that uses hybrid 1 only

- **SHOW 2** – Line is available for the show that uses hybrid 2 only
- **SHOW 1&2** – Line is available for both shows (shared line)

6. Press the SELECT button to accept your changes.

Note: You must reboot the system after configuring this setting. If you wish to configure lines as 'Priority' or 'Normal' (see below) you may do so before rebooting.

To configure lines as Priority or Normal lines:

1. Press the MENU softkey.
2. Press <∇> repeatedly until **Configure Shows** is highlighted and then press <SELECT>.
3. Enter the password and then press the OK softkey. NOTE: The default password is 1234.
4. Press <∇> repeatedly to highlight Set Priority Lines and then press <SELECT>.
5. Use <∇> and <Δ> to highlight each line. For each line, use the <+> or <-> Softkeys to select from the following options:
   - **Normal** – Line is included in the block-all group
   - **Priority** – Line is excluded from the block-all group. If this line is ringing, the Priority Ring Output (pin 2) on the parallel control port will go true.
6. Press the SELECT button to accept your changes.

Note: You must reboot the system after configuring this setting. Be sure to attach the phone lines before rebooting.

To change the password:

1. Press the MENU softkey
2. Press <∇> repeatedly until **Configure Shows** is highlighted and then press <SELECT>.
3. Enter the password and then press the OK softkey. **Note**: The default password is **1234**.
4. Press <∇> repeatedly to highlight **Change Password** and then press <SELECT>.
5. Enter a 4 digit password. Press each digit carefully. Then press the OK softkey. If you are uncertain if you have entered the desired password, press the BACK softkey instead of OK.
6. Press the BACK softkey repeatedly until back to the main menu.

Note: Restore Factory Defaults will restore the original password, but it will also erase all other settings.

Selecting a Show on the Desktop Director and other controllers

In the two-studio modes, each Desktop Director or other controller must select a Show.

Selecting a Show using a Desktop Director

To do so, follow these steps:

1. Press the MENU softkey
2. Scroll down until **Select Show** is highlighted and then press <SELECT>
3. Press <∇> or <Δ> arrow keys to select **Show 1** or **Show 2**.

Selecting a Show using a Call Controller

A given Call Controller can be associated with Show 1 or Show 2 by using a special key sequence. To assign a Console Director to a show, follow these steps:

1. All lines must be idle, or must be on hold, before you can change shows.
2. Press the following button sequence to associate the Console Director with Show 1: <*> <S> <H> <O> <W> <1> (or *74691).
3. Press the following button sequence to associate the Console Director with Show 2: <*> <S> <H> <O> <W> <2> (or *74692)

Selecting a Show using a Console Director

The Console Director is always in Talent mode. A Console Director can be associated with Show 1 or Show 2 by using a special key sequence. To assign a Console Director to a show, follow these steps:

1. When selecting a show, no line can be on the hybrid. All lines must be idle or must be on hold before you can change shows.
2. Press the following button sequence to associate the Console Director with Show 1 (Hybrid 1): <*> <S> <H> <O> <W> <1> (or *74691).
3. Press the following button sequence to associate the Console Director with Show 2 (Hybrid 2): <*> <S> <H> <O> <W> <2> (or *74692)

Connecting Assistant Producer to a Show

IMPORTANT

You must use Assistant Producer 3.5, or newer. If you are an Assistant Producer 2.0 user, contact Telos Customer Support for information on upgrading.

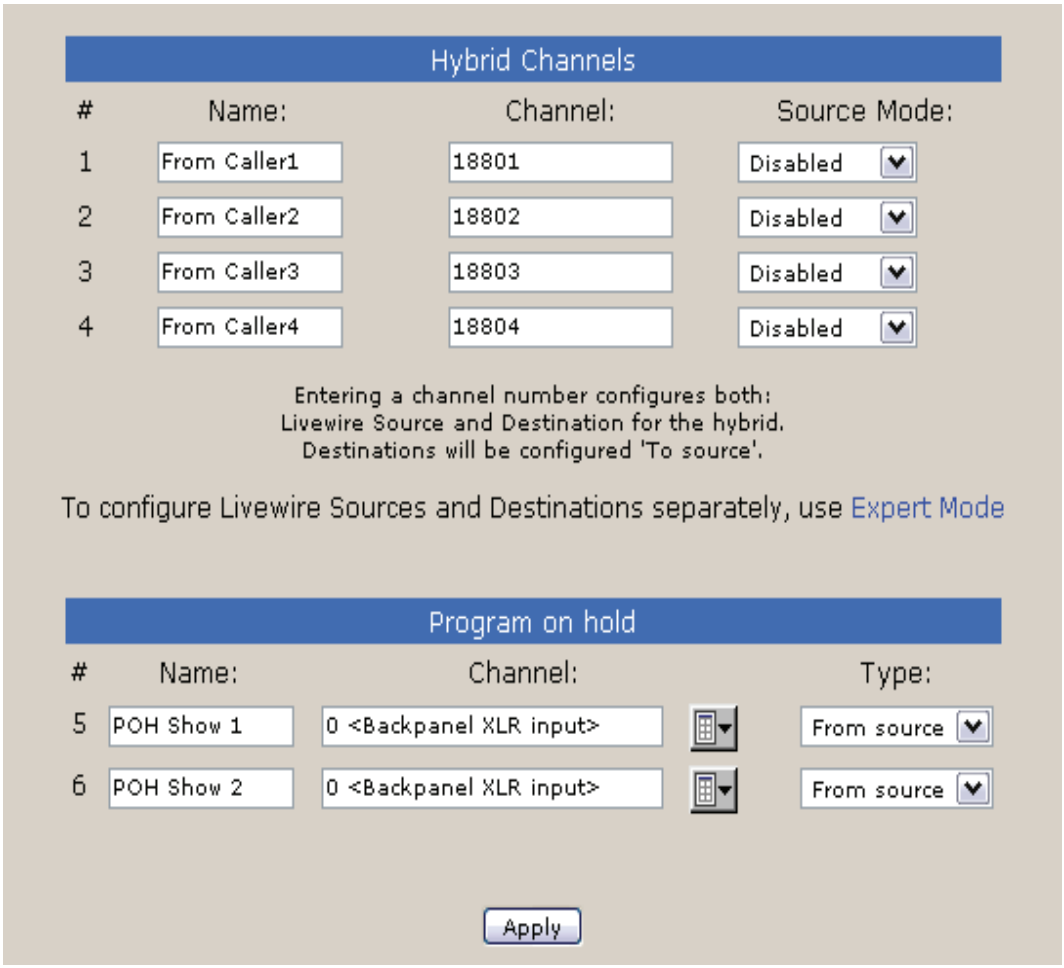
1. From the AP Server menu, choose a Server (the IP address of a Nx12)
2. From the AP Connection menu, choose Connect Show and from there choose from the desired show (Show 1 or Show 2)
3. From the AP Connection menu, choose Desktop Director and choose the Desktop Director you wish to control. NOTE: A Desktop Director's ID number is displayed on the upper left of the display. For example "DD2". To obtain the ID number of a Call Controller type <*><d><d> and it will be displayed on the screen.

For additional information, see the AP 3.5 User's Manual.

5.8 Livewire

The Web interface is used for Livewire configuration. The pages are similar to those for Livewire Nodes, so if you are already familiar with those, you will be immediately comfortable with the Nx12's Livewire configuration interface.

Consult *Introduction to Livewire*, available on the axiaaudio.com website for a description of Livewire and an explanation of the various terms and options. Note that the terms "Sources" and "Destinations" are from the perspective of the Livewire network and have the same usage as for Livewire Nodes.


Hybrid Channels
Name: Channel: Source Mode:
1 From Caller1 18801 Disabled
2 From Caller2 18802 Disabled
3 From Caller3 18803 Disabled
4 From Caller4 18804 Disabled
Entering a channel number configures both:
Livewire Source and Destination for the hybrid.
Destinations will be configured 'To source'.
To configure Livewire Sources and Destinations separately, use Expert Mode
Program on hold
Name: Channel: Type:
5 POH Show 1 0 <Backpanel XLR input> From source
6 POH Show 2 0 <Backpanel XLR input> From source
Apply

There are 4 hybrid channels. You can enter the Livewire text name and channel number or accept the defaults. Livewire channels are two-way, so this applies to both the send and receive audio. A Livewire enabled mixing console such as the Axia Element, when taking one of these as a source, will automatically connect both directions and provide the mix-minus. Cool, eh?

The two Program on Hold connections are Livewire "destinations", meaning that they are from the network to the Nx12. **Type** should normally remain at the default **From source** setting.

IMPORTANT

If you are not using a Livewire-ready network, do not enable Livewire streams or you will flood your Ethernet network. If you lose your Web access to the Nx12 after enabling the streams, this is probably what happened. (You can switch the streams back off by disabling Livewire from the main unit LCD Audio menu.) Only when a properly configured Livewire-compatible Ethernet switch is present should Livewire streaming be enabled.

When Livewire is enabled, the corresponding Nx12 Analog or AES3 inputs are switched-off. However, should a Livewire input fail; the Nx12 automatically switches the input to Analog/AES, providing a back-up.

On the other hand, the Nx12 analog/AES outputs will be simultaneously active when Livewire is enabled.

If you need to specify the source and destination Livewire channels independently, click the Expert Mode link to open a new page with these options.

Quality of Service

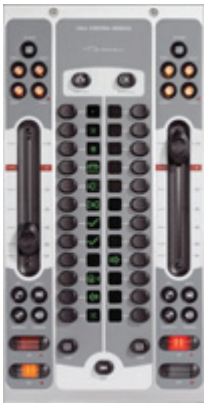
These are identical to the Livewire Node options. They should normally be left at the default values.

Axia Consoles

Axia mixing consoles are based on Livewire. This networked approach delivers many advantages, among them that Livewire connections are bi-directional, multi-channel, and include control. This means that a single RJ-45 can interface all necessary Nx12 signals to an Axia console. Livewire simplifies mix-minus because each channel is two-way – so return feeds are "baked-in" and not an afterthought.

Axia is a division of Telos, so as you could imagine, we take good care of phones and codecs in our mixing consoles. Easy-to-use mix-minuses are available on every fader channel. Separate on-air and off-line (recording) mix-minuses are supported. Each channel has a dedicated Talkback. There is a one-button off-line record mode.

And there are dedicated modules for phone control. These provide seamless integration, since they communicate with the Nx12 over the same Ethernet that is used for all other connections. Both USA-style operation with the well-known Telos two-column line selector approach and Euro-style with dedicated faders to each Telco line are available.

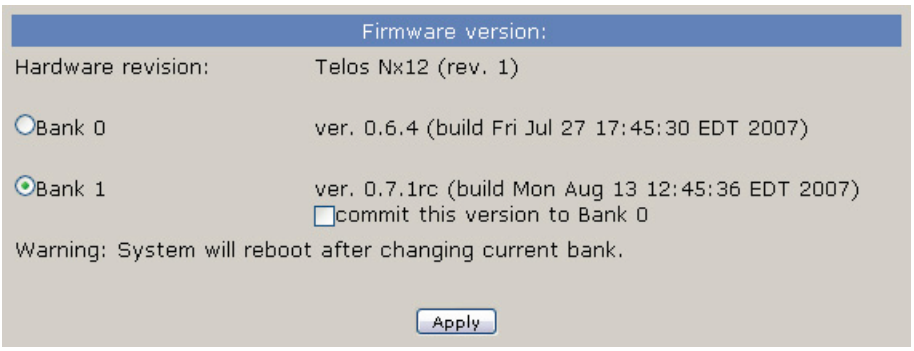


5.9 Firmware Update

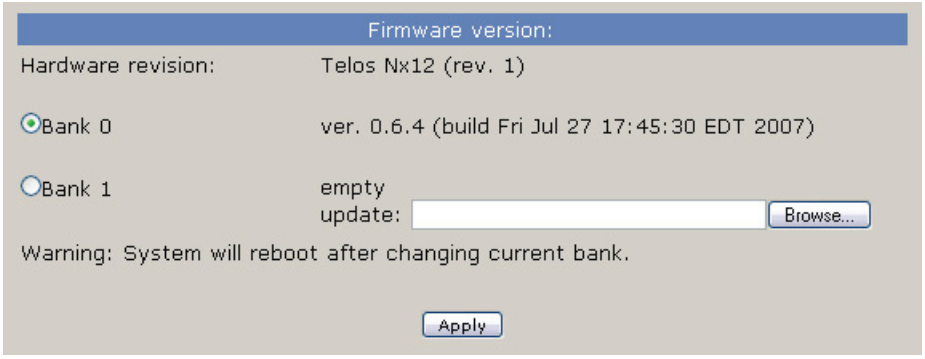
Nx12 firmware may be updated via the Web interface. Open your browser, connect to the Nx12, and click on the System link. Firmware update is at the bottom of the System page.

The first step is to get the new firmware onto your PC. This will usually be via a download from the Telos website or an email attachment from Telos customer support. The file will have the extension .pkg. When the firmware is available on your PC, click the Browse button to locate it, and then click **Apply** to upload it to the Nx12. The browser will disconnect for approximately 25 seconds, and then automatically re-connect. There will be a progress status screen, and then the System page will return. The new software will be in Nx12's Bank 1.

You can make it active by clicking the Bank 1 'radio button' and then **Apply**.


Firmware version:
Hardware revision:
Telos Nx12 (rev. 1)
Bank 0
ver. 0.6.4 (build Fri Jul 27 17:45:30 EDT 2007)
Bank 1
ver. 0.7.1rc (build Mon Aug 13 12:45:36 EDT 2007)
commit this version to Bank 0
Warning: System will reboot after changing current bank.
Apply

You can keep two firmware versions and choose between them at any time. But when you eventually want to upload a yet newer version, you will need to commit the firmware in Bank 1 to Bank 0, which clears Bank 1 to prepare it for the new upload. Do this by clicking the "commit this version to Bank 0" check box, followed by **Apply**.



5.10 Other Uses of the IP/Ethernet Interface

In addition to the Web user interface, the Ethernet port can be used to support call-control software applications and a telnet text-based remote user interface.

Call-control Software

Client software such as Telos' Assistant Producer and other 3rd-party applications such as NeoWinners from NeoSoft can connect to IP Port socket 9998 (decimal).

If you desire to write your own application to access the Nx12 via this protocol, please visit our website at www.telos-systems.com where you can download the specifications of this protocol and an SDK for Windows (see the downloads page section entitled 'Series 2101 development Kit').

Telnet

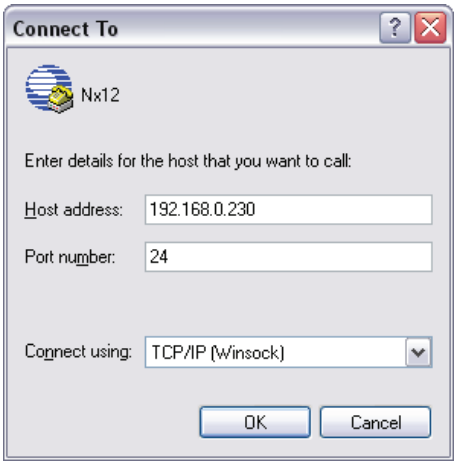
A telnet session can access certain advanced functions and logging, primarily for troubleshooting when requested by Telos support. Use your favorite telnet client to connect to the IP address and port 24 of the Nx12.

Some Web browsers support telnet: Entering 'telnet:192.168.0.230:24' into the browser's address field might work.

When starting a telnet session, a Login name is requested. The default is: **Telos**.

Windows Telnet

Since Windows comes with a simple telnet client, we will give you some tips on using this particular software. Run the HyperTerminal application at Accessories/Communications/HyperTerminal. Then from the File menu, select New Connection. Enter a name such as 'Nx12' and select an icon. When the window below appears, enter the Nx12's IP number, 192.168.0.230 (this is the Nx12 default – use your actual IP number, if you have changed it), enter '24' for Port Number, and select TCP/IP for the 'Connect using' field. Pressing OK will then take you to the telnet transaction screen.



Note that in the following section we will use bold type **like this** to indicate what you would type and Courier type like this to show what the Nx12 replies.

First you must log into the Nx12:

>>**Login Telos <return>**

```
Nx12  V0.6.3b, 2007-07-19 15:58

Logged in with user rights.
```

You should now be able to get the most up-to-date listing of commands by typing 'help' or '?':

>>**help <return>**

```
Type 'help <command>' to get specific help about a command.

Type 'help *' to get a short list of commands.

Type 'help **' to get a detailed list of commands.

Type 'help <chars>*' to get a list of commands starting with
<chars>.

'#' starts a decimal number, '%' a binary number, '$' a hex number.

Current number base is 16.

Separate multiple commands on a line by '::'.

List of available commands:

? CPC acc autoans bootstat bye call cidalert csinfo daa daacheck dd
disc

dmr dmw droponcptd fpkey gpio

help hookflash hybrid isdn isdnlog lawmode lcgood lcignore lcinfo
lcoffset

lcwait lines log login logon lw

mcheck meter minonhook mixpoll mr mw potscountry prio quit reboot
ringtimo

test ti tslot
```

For a description of a command, type 'help' or '?' followed by the name of a command. Example:

>>? CPC <return>

```
Command....: CPC <time in msec>

Description: Specify the time in msec a loop current drop must last
before a far-end drop is recognized. When set to 0 then loop cur-
rent drops are not detected at all.
```

A particularly useful command is for ISDN logging. This turns the Nx12 into an ISDN protocol analyzer so you can see the communications between the Nx12 and the Telco network.

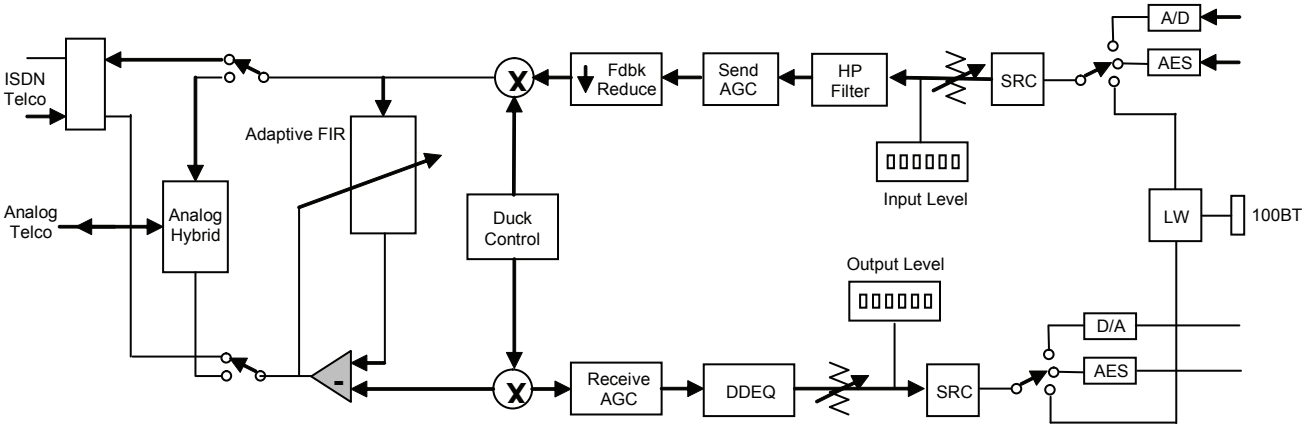
>>? Isdnlog <return>

```
? isdnlog

Command....: isdnlog <bitmask>

Description: Specify ISDN logging mode bitmask: b0=general logging,
b1=Layer2, b3=layer3 b4=IIEs, b5=raw packet, b6=Telesoft stack.
Usually, 0xF is a good choice.
```

5.11 System Signal Flow and Audio Processing



Send Audio Processing

The Nx12's send-to-caller audio processing consists of the following functions:

- Sample rate conversion
- High-pass filter
- Frequency shifter
- Send AGC/Limiter

Sample Rate Conversion

AES and Livewire sources are sample rate converted to the Nx12's internal sample rate of 8kHz. All inputs are re-clocked to the Nx12's internal clock (derived from the ISDN clock if at least one ISDN BRI is present).

High-Pass filter

A high pass filter with a 300Hz break frequency improves hybrid performance and enhances intelligibility by removing unnecessary low-frequencies from the input audio. This function is always enabled.

Frequency Shifter

A Telos innovation, the frequency shifter (pitch shifter) inserts a small, unnoticeable, shift in frequency to the send audio to prevent feedback buildup when the system is used with open speakers.

Send Automatic Gain Control / Limiter

This AGC helps maintain consistent audio levels to the caller. This function is always enabled. This benefits from a carefully-crafted algorithm utilizing a feed-forward topology with near in-audible artifacts below the limit threshold. At moderate levels it is 'AGC-like' while at high peak levels it is 'limiter-like'. In addition to making levels more consistent to callers, this performs the protection limiting required to meet Telecom regulations.

Receive (Caller) Audio Processing

The Nx12's receive audio processing consists of the following functions:

- High-pass "hum" filter

- Automatic Gain Control/Platform leveler
- Noise gate
- Dynamic equalization
- Sample rate conversion

High-pass 'hum' Filter

This filter removes hum and other unwanted low frequency noise from the caller audio. This filter has a break frequency of 100 Hz.

Receive Automatic Gain Control

This AGC serves to improve consistency of caller audio levels. This function is very important to effective hybrid performance as caller levels can vary by as much as 30 dB. The Nx12 offer a new approach to AGC to accommodate the fact that loud callers have become just as great a problem as low-level callers. The AGC is reset for each caller to a neutral position. to most-rapidly accommodate both low and high-level callers. The AGC has a range of up to +/- 12 dB. The AGC is an advanced dB- linear feed-forward topology providing a consistent sound independent of drive level. Its 'smart' operation normalizes levels while retaining the natural dynamics of the caller's voice. The default setting is FULL.

An important additional feature of this AGC is that it is cross-coupled to other sections of the Nx12 and can therefore reliably distinguish between caller audio and hybrid leakage. This allows a more aggressive gain control for bringing up low-level callers while still preserving excellent hybrid performance.

Noise Gate

Turning on the noise gate enables the built-in downward expander. The downward expander reduces low level line noise when no caller audio is present. And reduces low level leakage thereby improving hybrid performance. This function is cross-coupled with the AGCs and the ducking system.

Digital Dynamic Equalizer

Telephone audio frequency response varies widely as many different factors can effect it (we've measured the response on a number of calls and the results were revealing). Consequently, some form of receive equalization is desirable.

The Digital Dynamic EQ process used on the Telos is the most sophisticated equalizer available in a broadcast telephone interface. All processing is performed in the digital domain. The Receive EQ selection controls the type of equalization applied to the receive telephone audio as follows:

- **Off** - The caller audio is passed without modification.
- **Fixed** - This is a simple manual equalizer mode.
- **Adaptive** (Digital Dynamic) - This is a three band dynamic equalizer. We've chosen frequency breakpoints, time constants, and other characteristics to optimize the tonal quality of varied telephone callers. The user-entered EQ values are used as 'target levels'. This allows you to customize the caller spectral characteristics, but still get a consistent spectral characteristic from caller to caller.

Ducking System

The ducking function occurs primarily on the caller audio. This function is important for several reasons:

- Provides an "aesthetic" control over the caller that many programmers prefer. When the announcer speaks, the caller is ducked, or reduced in volume dynamically. This is particu-

larly useful when the caller is carrying on and the talent wants to interrupt to move the show along.

- Reduces feedback when a loudspeaker is necessary.
- In the loudspeaker case, reduces audibility of the caller signal that couples through the acoustic path, into the microphones, and is returned the caller resulting in a more natural sound for the caller.

The duplex system inserts a controlled loss (ducking) into whichever audio path (send or receive) is not active at the moment. When the caller is speaking, this loss is inserted in the announcer path, when the announcer is speaking the loss is inserted in the caller gain is reduced. The effect is somewhat "seesaw-like". If Feedback Reduction in the Audio menu is enabled, the Nx12 will increase ducking in the announcer-to-caller direction.

As a guideline, you will need more ducking when using open speakers. This helps reduce the occurrence of feedback and, also reduces the chances of the caller hearing himself via the speaker to mic path, which has an unnatural quality and can be disturbing. In a morning zoo type of scenario the setting should be even lower as there will typically be multiple mics used with open speakers. Fortunately, this scenario coincides with a more duplex style of operating – where pre-recorded 'bits' may be played to callers and must be heard without interruption. The default setting is 7.

You may choose how much of this effect you prefer, from Half Duplex (16dB) which makes the Nx12 operate like a one-way-at-a-time speakerphone, to Full Duplex (0dB) which disables ducking. A good starting value would be 6dB.

Feedback Control

We've carefully designed the Nx12 so that problems with feedback should be rare. However, despite excellent trans-hybrid loss and a number of other features, you can probably induce feedback if you try sufficiently hard, such as by cranking up your studio monitors to very loud levels. Some suggestions for what you should do if feedback does occur follow:

- Enable the Feedback Reduction function.
- Adjust Caller Ducking to a higher setting (more ducking).
- When mic processing is used, connect the hybrid in such a way that it gets an unprocessed mic signal.

  The problem here is that the mic processing combines with the Nx12's internal AGC to increase gain in the feedback path. Depending on the processor used, the feedback margin can be reduced by many dB. The Nx12's internal AGC has a internal adaptive smart-gate function to prevent inappropriate gain increase, but it is thwarted by this additional processing.

  If it is not possible to get an unprocessed mic signal, try to set the processor in such a way that room noise is not "sucked-up" during pauses. You can also try reducing the output the mic processor or set the Nom In (Nominal Input Level) selection in the Audio menu to the next higher setting, thereby reducing levels into the unit. The level to the caller should be OK since the Send AGC will compensate, however you will force that AGC towards its maximum gain, and thereby reduce how much gain it can add to the feedback path.
- Try repositioning the mics or speakers. It also helps, of course, to use directional mics. Customers have reported good results with cardioid mics such the EV RE-20 and the Shure SM-7.
- Add equalization to the monitor path. Acoustic resonances usually result in pronounced peaks in the feedback response. Since there are likely to be only a few pronounced peaks

in this response characteristic, flattening out those peaks with an equalizer can significantly improve the feedback margin. A spectrum analyzer connected to the output of your mic processor will help detect the frequency of those peaks.

- If necessary, soften acoustic room reflections by adding curtains or wall treatments.

The discussion above is intended to help in those situations where open speakers are a requirement. Whenever possible, of course, it is best to use headphones to listen to the caller audio. We have found the best scheme is to have the phone monitor speaker mute when the mic is turned on. If your mix minus has provided a means for the mic signal to be active even when the mic channel is turned off, the system can be used like a speaker phone when the taking calls off air. When the call is on the air the speaker mutes, however the talent will generally be wearing headphones when on mic, in any case.

6 Technical Data & Troubleshooting

6.1 Overview

A consequence of modern surface-mount construction is that it is frequently no longer possible for local repairs to be made. Special and expensive equipment is required to change parts. As well, today's equipment is very complicated and requires repair technicians to have detailed experience and training, and have access to high-end test equipment.

At the same time, the advent of overnight delivery services means that equipment can be returned to the factory for quick turn-around repair. Therefore, we do not expect you to fix this unit at the component level, and we do not include schematics of the unit in the manual. Upon request, our support staff can fax or mail you the needed schematic. Please see the first pages of this manual for proper procedures on returning units for repair.

There is the possibility, however, that you could repair the unit by swapping subsystems such as plug-in modules or power supplies. The Telos customer support crew is standing-by to assist with this, if you need it.

6.2 System Software & Firmware

Viewing Version Information

To assist with troubleshooting, Telos customer support may ask you to provide some software version numbers. There are two major software components to the system:

- System software – This is the software that can be updated by downloading new software.
- Firmware – This code is analogous to a bootstrap loader and contains certain critical low-level software. This information can only be updated or changed at the factory. It is stored in a partition of the non-volatile memory on the motherboard.

The software version and software build date can be determined from the start-up screen. The software version and software build-date are shown. (The build date is not the date your Nx12 was made.)

6.3 General Troubleshooting

Thinking about Problem Solving

Despite best intentions, something could always go wrong. Sometimes troubleshooting a balky set-up can make even the toughest engineer a Maalox and Rogaine addict.

You can't fix any system without the right world view; a zeitgeist of suspicion tempered by trust in the laws of physics, curiosity dulled only by the determination to stay focused on a single problem, and a zealot's regard for the scientific method. Perhaps these are characteristics of all who successfully pursue the truth. In a world where we are surrounded by complexity, where we deal daily with equipment and systems only half-understood, it seems wise to follow understanding by an iterative loop of focus, hypothesis, and experiment.

The notions here apply whether you are solving problems at the system level or at the component level. At the system level, the actions you might take would be very different – checking cables, trying different menu settings – but the thinking is the same.

Too many times, we fall in love with our suppositions. We are quick to overtly or subconsciously assume the problem being chased is due to lousy design, the stupid phone company, or the manager's latest memo.

Armed with a healthy skeptical attitude, the basic philosophy of troubleshooting any system is to follow these steps:

- Observe the behavior to find the apparent problem;
- Observe collateral behavior to gain as much information as possible about the problem;
- Round up the usual suspects;
- Generate a hypothesis;
- Generate an experiment to test the hypothesis;
- Fix the problem;
- Then, repeat, if necessary, to attack additional problems.

Let us now cover each step of the troubleshooting sequence in detail.

Step 1. Observe the behavior to find the apparent bug. In other words, determine the bug's symptoms. Remember always that many problems are subtle and exhibit themselves via a confusing set of symptoms.

Step 2. Observe collateral behavior to gain as much information as possible about the problem. Does the LCD's problem correlate to an LED flashing? Try to avoid studying a problem in isolation, but at the same time be wary of trying to fix too many at the same time. No one is smart enough to deal with multiple problems all at once – unless they are all manifestations of something more fundamental.

Step 3. Round up the usual suspects. At the system level, always suspect the menu set-up, the cables, the Phone Company line setup, the punch-blocks, etc. At the component level, many computer problems stem from the same sources. Never, never, never forget to check Vcc!

Step 4. Generate a hypothesis. Before changing things, formulate a hypothesis about the cause of the problem. You probably don't have the information to do this without gathering more data.

Sometimes you will have no clue what the problem might be. Sometimes, when the pangs of desperation set in, it's worthwhile to try anything practically at random. You might find a bad plug, an unconnected line, or something unexpected. Look around, but be always on the prowl for a working hypothesis.

Step 5. Generate an experiment to test the hypothesis. Change the ISDN connection to a known good line; call known good phone or hybrid at the other end; if long-distance doesn't work, try a local call.

You should plan your tests to eliminate 50% of the possible problems in one test, if possible. Just

keep careful track so you know what you have eliminated.

Step 6. Fix the problem.

A Final Thought…

Constantly apply sanity checks. More than three decades ago, the Firesign Theater put out an album called "Everything You Know is Wrong". Use that as your guiding philosophy in troubleshooting a Telos Two set-up. For example, just because you checked the Telco line last night and it was fine does not mean that it's OK now.

At 3:00 AM when the problems seem intractable and you are ready to give up engineering, remember that the system has worked and will work. Never panic–you are smarter than it is.

6.4 Troubleshooting ISDN Problems

Status Screen for ISDN troubleshooting

This screen shows the status of the ISDN connection. You can determine if the line is connected and whether the line and the Telos TWOx12 are successfully configured. In addition, information about calls is displayed here. One of the following status words will be shown for each active ISDN channel:

Deactivated
No line connected, or connected to wrong jack on the TWOx12. OR, ETS 300 permits the Telco to put a line to sleep. Deactivated will be displayed when this occurs

SPID Pending
The SPID has been sent to network (DMS Cust. And Natl I-1 only). Network has not approved SPID.

SPID Error
The SPID is incorrect or line is grossly misconfigured

Ready
The unit sees a live ISDN S interface (ie external NT1 is present or internal NT1 has completed startup). If the SPIDS are present, they have been sent and approved.

Pending Seize
Outgoing call in process; Request for call has been sent to Telco switch.

Seized
Outgoing call in process; Telco switch has acknowledged the call request.

Call Proceeding
Outgoing call in process; Telco switch waiting for routing info from network.

Outgoing Ring
Outgoing call in process; far end has generated a call alert signal

Disconnecting
Request to disconnect has been made. Awaiting acknowledgment.

Incoming Ring
An incoming call request has been detected and is being processed.

Accepting Call

Nx12 has accepted an incoming call and is waiting for the Telco switch to connect it.

Connected

All signaling is complete - the voice channel is available for use.

A caveat regarding the Ready indication: If you have this indication with the Nx12's Telco mode set to AT&T Cust and you cannot dial, it is likely that your line is Natl. I-1 or DMS-100. The AT&T Custom PTP protocol does not require SPIDs, so the Nx12 does not send or check them; it just assumes the line is OK in this regard, so it gives the ready indication, though the line is actually waiting for the SPIDs.

> **Note**
>
> **It is normal for Euro-ISDN ETS300 lines to go to the Deactivated state when not used. If the system can dial normally under these circumstances, this indication is not a cause for concern.**

Is the problem only with one BRI circuit or does it move when you switch lines around?

If the problem remains with the line, then call your Telco. On the other hand, if a line that works in one interface fails to work in another interface of the same Nx12 then you have an problem with your ISDN interface card (or possibly elsewhere in the Nx12).

Are you using the Nx12's internal NT1? (USA & Canada only)

The LED indicator of the status of the ISDN connection is on the rear panel near each of the the U jacks. If the circuit is inactive, the LED will remain in the rapidly flashing state. This state indicates a loss of the ISDN BRI circuit at the lowest (physical) level. Try another connecting cable or another line.

If the NT1 can contact the central office, the LED will blink slowly (about 1 Hz). If the LED remains in this state there is an internal ISDN problem or you could be connected to the wrong jack on the back (Users in USA & Canada should be using the lower, RJ11 style jack). The LED will come on solid when all handshaking is completed and the basic (low level) line connection is good.

If the LED stays on, with the ISDN disconnected, or does not come on at all after cycling power, there is a hardware problem.

Condition: Can't Get ISDN Ready Indication

First try disconnecting the ISDN circuit(s) in question and waiting a full 60 seconds (leave the Nx12 on) and then reconnect the circuit(s).

What kind of ISDN do you have, and is the Nx12 set to match it?

The telephone company should have given you the ISDN protocol when the line was ordered. If you find yourself in the field without this information, it is usually possible to piece together what you need.

- If you are given SPIDs, you probably have Natl ISDN 1 or DMS 100 for your ISDN protocol. One SPID means one channel was activated, rather than the usual two.
- No SPIDs and only one phone number generally means AT&T PTP protocol. You must not have SPIDs entered if using this protocol.
- In Europe, you have either ETS300 (EuroISDN) or the local protocol, such as ITR6 in Germany or VN2 in France. The Nx12 only works with ETS300.

In the case of Natl.I-1 or 2, and DMS-100 carefully verify that you have the correct SPID from the Telco, and that you have properly entered it into the system. It must be correct in order for anything to work! We've seen a few cases where it was given incorrectly by Telco personnel, so it might be a good idea to re-check with them if things are pointing in this direction.

Hot Tip

If you suspect the SPIDs given you are wrong, or the basic line provisioning (configuration) is incorrect, call the Telco and ask to speak to someone who can "log into the switch". They will probably need to have someone call you back. Only once you have this person on the line, and they are logged into the switch, ask them to look at the SPIDs programmed there. Also get this person's fax number and send them the ISDN ordering information form the Appendix of this manual.

Sometimes the information in the business office computer is wrong. By verifying what is programmed into the actual switch, you can save time. The AT&T PTP protocol generally does not allow two simultaneous voice calls. For this reason we recommend that the National ISDN-1 protocol be used if you are on the AT&T/Lucent 5ESS Telco switch.

We've included some useful diagnostic aids in the system to help you determine where a problem is originating. A good starting point is the ISDN status, accessible on the LCD by pressing any menu button twice.

Condition: System says Ready but attempted calls do not complete

(goes back to Ready or stays at Seize pending)

Are your SPIDs and Telco setting correct?

Can you dial from Line 1 to Line 2?

If you can successfully call from one line to the other, you know that your SPIDs and ISDN protocol settings for that line are correct and that the ISDN circuit itself is ok. Try a call to another local number.

If you cannot complete this call try dialing with a 9, 8, 7, or 6 before the number. Sometimes ISDN lines are set up as Centrex service even if you did not request this.

Is your long-distance carrier OK?

If you can successfully call locally, but long-distance calls don't go through, the Nx12 usually displays Far end disconnect, or No route available as the ISDN "Cause" phrase.

You may try another by dialing the appropriate 10XXXXX prefix (in the USA & Canada only).

In the USA you can determine who your long distance carrier is by dialing 1+700+555+4141. If Intra-LATA long distance competition has reached your area you can determine your Intra-LATA long distance carrier by dialing 1+area code+700+4141.

What does the ISDN "Cause" phrase say?

This phrase, which will appears on the ISDN protocol log after dialing, comes directly from the telephone company equipment, and can be valuable for troubleshooting. Generally, these phrases are self-explanatory. Getting them means you are at least talking to the Central Office equipment. There is a full description of the phrases in Section 6.7.

Condition: Intermittent ISDN Problems

These are among the most difficult kinds of problems to solve. Rest assured, however, that we have yet to lose one!

First try disconnecting the ISDN line and waiting a full 60 seconds (leave the Nx12 on) and then reconnect the line.

Does the problem occur on all of the ISDN BRI circuits or just a single BRI circuit (2 "lines")?

If the problem is limited to a single BRI circuit (ie both "lines" on that BRI are showing the problem) then the problem may be with that BRI line or with that ISDN interface card. Move the BRI line in question to a different interface card (don't forget to reprogram the SPIDs). If the problems moves it must be with the BRI itself. If it remains with certain "lines" on the system then it is probably in the ISDN interface card. Swapping cards between two of the interface slots will confirm or disconfirm this.

If the NT1 sync light for a given ISDN BRI circuit occasionally cycle to the fast flashing state, then the line or NT1 is marginal.

If the problem occurs on 2 BRI circuits simultaneously, and they are both in the same interface slot, then it is likely the problem is with that ISDN interface card. Swapping cards between two of the interface slots will confirm or disconfirm this.

Are the problem confined to incoming call only?

In this case, check the ISDN (Telco) protocol setting in the TEL. If you believe your line is DMS Custom try the National ISDN setting, or vice versa.

Are the DNs entered correctly? If so, try using 10 digits rather than 7 digits, or vice versa.

What is consistent about the problem (look for patterns)?

Keep watching for a pattern. Enlist the help of the users. We've seen apparently random problems that we eventually discovered only occurred when it rained, or on a certain day each week, or at a certain time of day, or only when another line was in use. In other cases, it was related to temperature or dirty AC power. Assume nothing and suspect everything. Leave no stone unturned while searching for the answer.

Any other ISDN troubleshooting tips?

An important tool for troubleshooting incompatibility/programming problems is the Nx12's integrated ISDN analyzer. This enables you to look at the transactions between the Telco line and the Nx12, so that a judgment can be made as to whether the problem lies within the Nx12 or with the programming of the Telco switch.

The usual cause of inability to dial problems is a Telco problem, such mis-programming.

If you are convinced that the problem lies with the ISDN line configuration, you may be able to convince the Telco to have an engineer look at the line with an analyzer. This works much like the Nx12's internal analyzer, but with more detail, and in a format that is more readily understandable by telephone people. This procedure may sometimes be the only way to get attention to a line problem. It is hard to deny evidence provided by their own gear.

In rare cases, there could be a software bug in the Nx12 that reacts badly to something coming from the ISDN line. Telos customer support will communicate the problem to our engineering guys for resolution. See Section 7.41 for information on connecting to the Nx12's command line

to turn on ISDN logging.

6.5 Troubleshooting Desktop Director Problems

Most problems with the Desktop Directors are cable related. First try a new cable. If that fails to solve the problem, try the Desktop Director with a short cable plugged directly into each of the Director Ports on the back of the Nx12.

Note that early production Call Controllers require a modification to operate with the Nx12. Telos customer support can arrange for modification of these units at no charge.

Section 2 has details on the allowable wiring configurations for the Desktop Directors and other control surfaces. There are distance and wiring restrictions. Note that when two Directors share a single port an external power supply is required and the termination resistor must be removed on the unit closer to the power supply.

The diagnostics on the Desktop Director and the Nx12's status screen will help isolate bad wiring or hardware. If errors are accruing try systematically swapping ports, and swapping Desktop Directors, to see if the problem stays with a given port, Desktop Director, or cable.

6.6 Replaceable Modules and Hardware Servicing

WARNING - DANGER

Remove all connections, including the Telephone, Audio, and Electrical Mains Power cord before opening the unit. Failure to do so could result in injury or death. Do not open the unit unless you are qualified to do so.

Removal of the top plate gains access for service. To do so, remove the 10 Philips head screws.

Replaceable modules

While we do not expect you to do much repair or troubleshooting of the individual circuit boards, the system does have several removable modules which can be replaced. The system has the following replaceable modules:

1. Power Supply
2. ISDN Line interface card:

   Part # 1701-00140 (U interface - USA and Canada); or

   Part # 1701-00139 (S interface – Worldwide)
3. Analog POTS Line Interface part # 1701-00141
4. AES option card part # 1701-00149

Fuse Replacement

A fuse is located on the power supply PCB. If fuse replacement is required, note: **For continued protection against fire, replace fuse only with same type and value.**

Access to the Main PCB

The main PCB, closest to the bottom of the enclosure, is mounted component-side down. You can gain access to the component side of the PCB without taking it out by removing the bottom plate of the enclosure box.

6.7 ISDN Cause Phrases/Values

When there is an ISDN problem in the network, a phrase appears in the ISDN protocol log that communicates information about the cause of the trouble. See Section 7.10 for how to enable ISDN logging through the Telnet port. These "Cause Values" are numbers generated by the network. When possible, we use those suggested by the Bellcore standard. Note that in some cases there may be more than one meaning. This can frequently be evaluated by whether the message has been received by the calling party or the called party.

CLASS 1.1 (normal)

Cause No. 1 - Check number, redial (unallocated number)
This cause indicates that the called party cannot be reached because, although the called party number is in a valid format, it is not currently allocated (assigned).

Cause No. 2 - No route to network /Prefix 0 dialed in error
This cause indicates that the equipment sending this cause has received a request to route the call through a particular transit network which it does not recognize. The equipment sending this cause does not recognize the transit network either because the transit network does not exist or because that particular transit network, while it does exist, does not serve the equipment which is sending this cause.

or

This cause indicates the prefix 0 is invalid for the entered number

Cause No. 3 - No route to dest./Prefix 1 dialed in error
This cause indicates that the called party cannot be reached because the network through which the call has been routed does not serve the destination desired. This cause is supported on a network-dependent basis.

or

A 1 was dialed when not required. Redial without the 1.

Cause No. 4 – No prefix 1/Special Information Tone
The prefix 1 is not required for this number

or

This cause indicates that the called party cannot be reached for reasons that are of a long term nature and that the special information tone should be returned to the calling party.

Cause No. 5 – Misdialed trunk prefix (National use)
This cause indicates the erroneous inclusion of a trunk prefix in the called party number.

Cause No. 6 – Channel unacceptable
This cause indicates a called user cannot negotiate for a B-channel other than that specified in the SETUP message.

Cause No. 7 – Call awarded and being delivered on an established channel
This call indicates the user has been awarded the incoming calls, and that call is being connected to a channel already established to that user for similar calls (e.g. packet-mode, X.25 virtual calls, etc).

Cause No. 8 – Call is proceeding/Preemption
Call in process, please standby.

or

This cause indicates a call has been preempted.

Cause No. 9 – Preemption, reserved
This cause indicates a calls has been preempted due to the circuit being reserved for reuse.

Cause No. 10 – Prefix 1 not dialed [DMS Custom only]
This cause indicates that the prefix 1 was not included in the called party address and is required.

Cause No. 11 – Excessive Digits Received, call is proceeding
More digits were dialed than expected. Called number has been truncated to the expected number.

Cause No. 14 – Excess digits received, call is proceeding
More digits were dialed than expected. Called number has been truncated to the expected number.

Cause No. 16 - Far end disconnect (Normal call clearing)
This cause indicates that the call is being cleared because one of the users involved in the call has requested that the call be cleared. Under normal situations, the source of this cause is not the network.

Cause No. 17 - Busy, try again later (User busy)
This cause is used to indicate that the called party is unable to accept another call because the user busy condition has been encountered. This cause value may be generated by the called user or by the network. In the case of user determined user busy it is noted that the user equipment is compatible with the call.

Cause No. 18 - No far end response (No user responding)
This cause is used when a called party does not respond to a call establishment message with either an alerting or connect indication within the prescribed period of time allocated (before timer T303 or T310 has expired).

Cause No. 19 - No answer (No answer from user, user alerted)
This cause is used when the called user has provided an alerting indication, but not a connect indication within a prescribed period of time (before timer T301 has expired).

Cause No. 20 – Subscriber absent
This cause is used when a mobile station has logged off, radio contact is not obtained with a mobile station or a personal telecommunications user is temporarily not addressable at any user-network interface.

Cause No. 21 - Call rejected
This cause indicates that the equipment sending this cause does not wish to accept this call, although it could have accepted the call because the equipment sending this cause is neither busy nor incompatible.

This cause may also be generated by the network, indicating that the call was cleared due to a supplementary service constraint.

Cause No. 22 - Number changed
This cause is returned to a calling party when the called party number indicated by the calling

party is no longer assigned, The new called party number may optionally be included in the diagnostic field. If a network does not support this cause, cause no: 1, unallocated (unassigned) number shall be used.

Cause No. 26 – Non-selected user clearing
This cause indicates the user has not been awarded the incoming call.

Cause No. 27 – Destination out of order
This cause indicates that the destination indicated by the user cannot be reached because the interface to the destination is not functioning correctly. The term "not functioning correctly" indicates that a signal message was unable to be delivered to the remote party; e.g. a physical layer or data link layer failure at the remote party, or user equipment off-line.

Cause No. 28 - Incorrect number (invalid number format, address incomplete)/Special intercept announcement
This cause indicates that the called party cannot be reached because the called party number is not in a valid format or is not complete.

or

This cause indicates the user should be returned a Special Intercept Announcement.

Cause No. 29 – Facility rejected/Special Intercept announcement: undefined code
This cause indicates a facility requested cannot be provided by the network.

or

This cause indicates that a user in a special business group (i.e. Centrex) dialed an undefined code.

Cause No. 30 – Result of a STATus ENQuiry/Special intercept announcement: number unassigned
This cause is included in the Status Message when the reason for sending the Status Message was previous receipt of a Status Enquiry message.

or

This value indicates that a user from outside a basic business group (i.e. Centrex) has violated an access restriction feature.

Cause No. 31 - Network disconnect (Normal, unspecified)/Special intercept announcement: Call blocked because of group restricitons
This cause is used to report a normal event only when no other cause in the normal class applies.

or

This value indicates that a user from outside a basic business group (i.e. Centrex) violated an access restriction feature

CLASS 1.2 (resource unavailable)

Cause No. 34 - No circuit available (circuit/channel congestion)
This cause indicates that there is no appropriate circuit/channel presently available to handle the call.

Cause No. 35 – Call Queued [5ESS Custom only]
This indicates the call has been queued.

Cause No. 38 - Net out of order

This cause indicates that the network is not functioning correctly and that the condition is likely to last a relatively long period of time e.g. immediately re-attempting the call is not likely to be successful.

Cause No. 39 – Permanent frame mode connection is out-of-service [Q.850 only]

This cause indicates a calls has been preempted due to the circuit being reserved for reuse.

Cause No. 40 – Permanent frame mode connection is operational [Q.850 only]

This cause indicates a calls has been preempted due to the circuit being reserved for reuse.

Cause No. 41 – Net problem, redial (Temporary Failure)

This cause indicates that the network is not functioning correctly and that the condition is not likely to last a long period of time; e.g. the user may wish to try another call attempt almost immediately. May also indicate a data link layer malfunction locally or at the remote network interface or that a call was cleared due to protocol error(s) at the remote network interface.

Cause No. 42 – Net busy, redial (Switching Equipment Congestion)

This cause indicates that the switching equipment generating this cause is experiencing a period of high traffic.

Cause No. 43 – Access/user information discarded

This cause indicates that the network unable to deliver user information (i.e user-to-user information, low level compatibility or sub-address, etc) to the remote users as requested.

Cause No. 44 – No channel available (Requested circuit/channel not available)

This cause is returned when the circuit or channel indicated by the requesting entity cannot be provided by the other side of the interface.

Cause No. 46 – Precedence call blocked [Q.850 only]

This cause is used to indicate that there are no pre-emptable circuits or that the called user is busy with a call of equal or higher pre-emptable level.

Cause No. 47 – Resource unavailable/New Destination

This cause is used to report a resource unavailable event only when no other cause in the resource unavailable class applies.

or

This cause is used to indicate that the original destination is unavailable and to invoke redirection to a new destination.

Cause No. 49 – QOS unacceptable

This cause is returned when the Quality of Service is unacceptable.

CLASS 1.3 (service or option not available)

Cause No. 50 – Requested facility not subscribed

The cause is used to report that the user cannot use this feature because s/he has not subscribed to it.

Cause No. 51 – Bearer capability incompatible with service request

This cause indicates a user request for action was rejected because the action was incompatible with the capability of the call.

Cause No. 52 – Outgoing calls barred

This cause indicates that because of call screening provided by the network, the calling user is not permitted to make a call.

Cause No. 53 – Service operation violated

This cause indicates that the user has violated the service operation.

Cause No. 54 – Incoming calls barred

The user will not accept the call delivered in the SETUP message.

Cause No. 55 – Incoming calls barred within CUG at destination

This cause indicates that incoming calls are not permitted to members if this Closed User Group

Cause No. 57 – Bearer capability (Data/voice) not authorized.

This cause indicates that the user has requested a bearer capability which is implemented by the equipment which generated this cause but the user is not authorized to use it. This is a common problem caused by wrong Telco provisioning of the line at the time of installation.

Cause No. 58 – Bearer capability not presently available

This caused indicates the users has requested a bearer capability which has been implemented by the equipment, but is not available at this time.

CLASS 1.4 (service or option not implemented)

Cause No. 62 – Inconsistency in designated outgoing address information/Service Not Authorized [VN4]

This cause indicates an inconsistency in designated outgoing address information and subscriber class.

or

This cause indicates that a Terminal Diversion in a VN4 network has been rejected.

Cause No. 63 – Service or option not available, unspecified

This cause is used to report a service or option not available, only when no other cause in this class applies.

Cause No. 65 - Bearer Capability not implemented (Incompatible bearcap)

This cause indicates that the equipment sending this cause does not support the bearer capability requested.

Cause No. 66 – Channel type not implemented

This cause is returned when the called party has reached a channel type not supported.

Cause No. 69 – Requested facility not implemented

This cause indicates that the network (or node) does not support the requested bearer capability and therefore cannot be accessed at this time.

Cause No. 70 - Restricted only

This cause indicates that the calling party has requested an unrestricted bearer service but the equipment sending this cause only supports the restricted version of the requested bearer capability.

Cause No. 79 – Service or option not implemented, unspecified

This cause is used to report a service or option not implemented event only when no other cause in this class applies.

CLASS 1.5 (invalid message; e.g. parameter out of range)

Cause No. 81 – Invalid call reference value

This cause indicates that the equipment sending this cause has received a message with a call reference which is not currently in use on the user-network interface. This value only applies if the call reference values 1 or 2 octets long and is not the global call reference.

Cause No. 82 – Identified channel does not exist

This cause is sent when the equipment sending this cause has received a request to use a channel not active on the interface for a call

Cause No. 83

A suspended call exists, but this call's identity does not

Cause No. 84 – Call identity in use

This cause indicates that a call identity is in use.

Cause No. 85 – No call suspended

This cause indicates that no call is suspended.

Cause No. 86 – Requested call identity cleared

This cause indicates that the call having the requested called entity has cleared.

Cause No. 87 – Destination not a member of CUG

This cause indicates that the address of an outgoing call attempt is not a member of the Closed User Group.

Cause No. 88 - Incompatible destination

This cause indicates that the equipment sending this cause has received a request to establish a call which has low layer compatibility, high layer compatibility or other compatibility attributes (e.g. data rate, DN subaddress) which cannot be accommodated. This call can also be returned by a switch to a CPE when trying to route a call to an incompatible facility, or one without data rate.

Cause No. 90 – Destination address incomplete/Non-existent CUG

This cause indicates that the destination address is missing or incomplete

or

Indicates a non-existent Closed User Group

Cause No. 91 - Invalid transit network selection

This cause indicates that an Invalid transit network selection has been requested.

Cause No. 92 – Invalid supplementary service parameter [VN4 only]

Used to indicate an invalid parameter for supplementary service

Cause No. 95 – Invalid message, unspecified

This cause indicates that the entity sending this cause has received and Invalid message, only when no other cause in this class applies.

CLASS 1.6 (protocol error; e.g. unknown message)

Cause No. 96 – Mandatory Info missing

This cause indicates that the equipment sending this cause has received a message which is missing an information element which must be present in the message before that message can

be processed.

Cause No. 97 - Message type non-existent or not implemented

This cause indicates that the equipment sending this cause has received a message with a message type it does not recognize either because this is a message not defined, or defined and not implemented by the equipment sending this cause.

Cause No. 98 – Message not compatible

This cause indicates that the message received is not compatible with the call state or the message type is non-existent or not implemented.

Cause No. 99 – Info non-existent or not implemented, call processed

This cause is sent when the equipment sending this cause has received a message which includes the information elements not recognized because the information element identifier is not define or it is defined but not implemented by the equipment sending the cause. However, the information element is not required for the equipment sending the cause to process the message.

Cause No. 100 – Invalid information element contents

This cause is returned when the equipment sending this cause has received an information element that it has implemented; however one or more field of the information element are coded in such as way (e.g., truncated, invalid extension bit, invalid field values, etc) that has not been implemented by the equipment sending this cause.

Cause No. 101 – Message not compatible with call state/Protocol error threshold

This cause is returned when the equipment sending this cause has received a message that procedures indicate is not a permissible message to receive at this time.

or

This cause indicates that the switch sending this cause is clearing the call because a threshold has been exceeded for multiple protocol errors during an active call.

Cause No. 102 - Timeout disconnect (Recovery on timer expiration)

This cause indicates that a procedure has been initiated by the expiry of a timer in association with error handling procedures.

Cause No. 103 – Parameter non-existent or not implemented – passed on (National use)/ Mandatory Information Element of incorrect length [VN 4 only]

This cause indicates that the equipment sending this cause has received a message which includes parameter(s) not recognized because the parameter(s) are not defined of are defined but not implemented by the equipment sending this cause. This cause indicates that the parameter(s) were ignored. In addition, if the equipment sending this cause is an intermediate point, then this cause indicates that the parameter(s) were passed unchanged.

or

This cause indicates that the equipment sending this cause has received a message with a mandatory information element of incorrect length.

Cause No. 110 – Message with unrecognized parameter discarded [Q.850 only]

This cause indicates that the equipment sending this cause has discarded a received message that includes a parameter that is not recognized.

Cause No. 111 – Protocol Error, unspecified

This cause is used to report a protocol error event only when no other cause in this class applies.

Among other things, this cause may be displayed if you failed to dial a “9” or “8” for an outside line, if required. Also may be returned if you have some types of restrictions as to the number of

calls, etc.

Cause No. 113 – Bearer service unavailable [VN 4 only]

This cause indicates that the bearer service requested is not available or is unimplemented.

Cause No. 114 – End-to-End Information transfer impossible [VN 4 only]

This cause indicates that the requested network does not support the user end-to-end transfer capability.

CLASS 1.7 (interworking class)

Cause No. 126 – Switch to conversation mode [VN 4 only]

This cause is used to indicate that conversation mode has begun.

Cause No. 127 - SW56 disconnect (Internetworking, unspecified)

This cause indicates that an interworking call (usually a call to SW56 service) has ended. May also be seen in the case of a non specific rejection by your long distance carrier (try again at a different rate)

7 Specifications

General

Interface for connecting six ISDN BRI (SØ) or twelve analog telephone lines to professional studio equipment. Includes four high-performance digital hybrids and a digital matrix switch. All hybrids can be dedicated to a single studio for excellent conferencing results, or the Nx12 may be used to serve two studios, with two hybrids each.

Audio interfaces included as standard are analog and Livewire Audio-over-IP. An optional AES interface module is available.

Up to eight Desktop Directors may be connected. Computer control, such as with Telos' Assistant Producer call screening application, is supported over an Ethernet LAN connection.

Processing Functions

General

- Telos 3rd-generation adaptive digital hybrids
- Telos exclusive feedback reduction functions, including Acoustic Echo Cancellation

Send (to caller) Processing

- Sample rate conversion
- High-pass filter
- Frequency shifter
- AGC/limiter
- Pgm-on-Hold AGC/limiter
- Sample rate conversion

Receive (from caller) Processing

- High-pass "hum" filter
- Smart AGC/platform leveler
- Noise gate
- Telos' DDEQ (Digital Dynamic Equalization) 3-band adaptive spectral control processor
- Sample rate conversion

Audio

Analog Inputs

- 2x send analog inputs
- 2x Program-on-Hold analog inputs
- XLR female, pin 2 high. Active balanced, with RF protection.
- **Output level** adjustable from –7 to +8 dBu (nominal)
- **Analog clip point** +21dBu
- **Output impedance** <60 Ohms x2
- **Analog-to-digital converter resolution** 20 bits

Analog Outputs

- 2x receive analog outputs
- XLR male, pin 3 high.
- **Input level** Adjustable, -7 to +8 dBu (nominal) level.
- **Impedance** Bridging, >100K Ohms
- **Digital-to-analog converter resolution** 24 bits
- **Headroom before clipping** 20 dB from +4dBu nominal level

AES3 Digital Inputs/Outputs (option)

The AES input/output option module substitutes 2x input and 2x output AES XLRs in place of the standard analog, providing 4x in/4x out channels.

- Conform to AES3 standard
- **Sample rates** 32, 44.1, and 48kHz.
- **Rate conversion** Input and output, independently selectable
- **Clock** AES input or ISDN network clock.
- **Input Level** Adjustable -27 to –12 dBfs
- **Output Leve**l Adjustable –27 to –12dBfs

Livewire

- Livewire connection is standard for all hybrids in/out and POH audio in

Audio Performance

Frequency Response ± .5dB, 50 to 20kHz (swept sine procedure, measured from analog input to output with unit in loopback mode)

THD+N/Input <0.06% typical (Measured at 0 dBu @ 1 kHz analog in to AES out in loop-back mode)

Switching Matrix and Conferencing

- All-digital audio routing and switching

- 12 telephone lines, 4 hybrids, 4 studio inputs, 4 studio outputs, 2 Program-on-Hold inputs

ISDN Telephone Line Connectivity

Protocol Compatibility:

- National ISDN-1 and 2
- DMS-100 Custom Functional
- AT&T 5ESS Custom Point-to-Point
- Euro-ISDN conforming to the NET 3/ETS300 protocol

Interface (one of the following)**:**

- **USA & Canada** Integrated NT1 for direct connection to ISDN line via the 2-wire U-interface (6-position/2-pin RJ-11 connector). 2B1Q line coding.
- **Worldwide** 4-wire S interface (8-position/8-pin RJ-45 connector)

Telephone Coding Modes

- μLaw (ISDN Proto set to Natl I-1, AT&T Custom, Q.931mu or DMS Custom)
- A-Law (ISDN Proto set to ETS-300)

Analog Telephone Line Connectivity

Universal interface for worldwide application. Programmable loop current, ring signaling, and flash time. Includes caller ID decoding using Bellcore 212 modem standard.

Desktop Director Ports

Four ports, permitting connection of 4 Telos Desktop Directors. Eight directors using external RJ-45 splitters and power. (Extended Desktop Directors count as 2 for power purposes.)

Control Ports

Ethernet 100Base-T

- Web server for configuration and software updates
- Telnet for command line control and diagnostics
- Assistant Producer server allows up to 8 instances of Telos Assistant Producer (version 3.5 or later) to connect simultaneously

General purpose Input/Output

15 pin D-Sub connector with 5 status outputs and 4 control inputs.

Two-Studio Modes

The Nx12 can be used with two studios independently. Lines may be independent or shared. Each studio has two hybrids. There are two Progam-on-Hold inputs, one for each studio. Two sets of Telos' Assistant Producer application can be used independently.

Note the following:

- With analog audio in/out, there are only two total inputs and outputs. So the Nx12 must be used in a mode that combines the two hybrids assigned to each studio into one input/output.
- The AES option provides 4 input and 4 output channels total, so all hybrids are accessible with each studio using one AES connection's two channels. Program-on-Hold remains an analog input.
- Livewire provides the full 4 hybrid input channels, 4 hybrid output channels, and 2 Program-on-Hold inputs.

Four-Column/Four-Hybrid Mode

The Nx12 can be used with an extended Desktop Director, which has four telephone line columns. Each has an associated hybrid.

Note the following:

- With analog audio in/out, there are only two total inputs and outputs. So the Nx12 must be used in a mode that combines the hybrids into two groups of two each.
- The AES option provides 4 input and 4 output channels total, so all hybrids are accessible. Program-on-Hold remains an analog input.
- Livewire provides the full 4 hybrid input channels, 4 hybrid output channels, and a Program-on-Hold input.

"Euro" Operation Mode

European broadcasters traditionally prefer to have an independent signal path from each telephone line, through an independent hybrid, and to a mixing console fader dedicated to each. No line selection matrix switch is used. The Nx12 supports this operation style with Axia mixing consoles for up to four lines. A standard Desktop Director is used for off-air conversation and call screening.

8 Warranty and Application Caution

TELOS Nx12 LIMITED WARRANTY

This Warranty covers "the Products," which are defined as the various audio equipment, parts, software and accessories manufactured, sold and/or distributed by TLS Corp.,d/b/a Telos Systems (hereinafter "Telos Systems").

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of one year from the date of receipt by the end-user. Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the date of receipt by the end-user.

This warranty is void if the Product is subject to Acts of God, including (without limitation) lightning; improper installation or misuse, including (without limitation) the failure to use telephone and power line surge protection devices; accident; neglect or damage.

EXCEPT FOR THE ABOVE-STATED WARRANTY, TELOS SYSTEMS MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will Telos Systems, its employees, agents or authorized dealers be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

In order to invoke this Warranty, notice of a warranty claim must be received by Telos Systems within the above-stated warranty period and warranty coverage must be authorized by Telos Systems. If Telos Systems authorizes the performance of warranty service, the defective Product must be delivered, shipping prepaid, to: Telos Systems, 2101 Superior Avenue, Cleveland, Ohio 44114.

Telos Systems at its option will either repair or replace the Product and such action shall be the full extent of Telos Systems' obligation under this Warranty. After the Product is repaired or replaced, Telos Systems will return it to the party that sent the Product and Telos Systems will pay for the cost of shipping.

Telos Systems' authorized dealers are not authorized to assume for Telos Systems any additional obligations or liabilities in connection with the dealers' sale of the Products.

Telos products are to be used with registered protective interface devices that satisfy regulatory requirements in their country of use.

A1 USA SPIDs by Telco

Here is a list of known working SPIDs (Service Profile IDs) by telephone company, compiled from our direct experience helping clients with ISDN installations. Your SPIDs may be different, of course.

The SPID is distinct from your telephone number, and does not necessarily contain your area code or telephone number, although this is generally the case. Incorrect SPIDs will be indicated by the line status indication **Wait** on the Nx12 ISDN status screen. Often the Telco will leave the last "00" or "01" off the end of your SPID when giving it to you. For the latest list check Telos Systems' web page.

A standardized SPID format for national ISDN of XXXYYYZZZZ0101 was phased in during 1996 and is used by most of the Telcos for new lines.

In the list below, XXX=Area code; YYY=exchange; ZZZZ=phone number.

Alltel (Ohio)

XXXYYYZZZZ0101

Ameritech

XXXYYYZZZZ0111

or

XXXYYYZZZZ011

or

01YYYZZZZ011

or

XXXYYYZZZZ01

Bell Atlantic

01YYYZZZZ00

or

01YYYZZZZ000

or

01XXXYYYZZZZ000

or

XXXYYYZZZZ100

Bell Atlantic North (Former NYNEX)

XXXYYYZZZZ0101

or

XXXYYYZZZZ0000

Bell Canada

XXXYYYZZZZ00

or

XXXYYYZZZZ0101

BellSouth

XXXYYYZZZZ0101

or

XXXYYYZZZZ0100

or

XXXYYYZZZZ01

or

01YYYZZZZ0

or

01YYYZZZZ000

Century Telephone
XXXYYYZZZZ00

Cincinnati Bell
01YYYZZZZ000
or
SPID1 00YYYZZZZ01
SPID2 00YYYZZZZ02
or
00YYYZZZZ01

Fort Mills Telephone
XXXYYYZZZZ

Frontier Communications
01YYYZZZZ000

GCI (Alaska)
XXXYYYZZZZ00

GTE
XXXYYYZZZZ0101
or
XXXYYYZZZZ0001
or
01YYYZZZZ000
or
XXXYYYZZZZ00
or
01YYYZZZZ0
or
XXXXYYYZZZZ
or
SPID1 XXXYYYZZZZ00
SPID2 XXXYYYZZZZ01
or
SPID1 XXXYYYZZZZ0000
SPID2 XXXYYYZZZZ0001
or
SPID1 XXXYYYZZZZ0000
SPID2 XXXYYYZZZZ0101

Nextlink/XO
XXXYYYZZZZ
or
XXXYYYZZZZ1
or
XXXYYYZZZZ0101 (if requested)

Northern Pittsburgh Telephone
XXXYYYZZZZ000

Pacific Bell
XXXYYYZZZZ0101
or
SPID1 XXXYYYZZZZ01
SPID2 XXXYYYZZZZ02
or
01YYYZZZZ000
or
SPID1 XXXYYYZZZZ1
SPID2 XXXYYYZZZZ2
or
XXXYYYZZZZ00
or
XXXYYYZZZZ

Quest- See US West

Rochester Telephone

01YYYZZZZ0

SBC- See Pac Bell, Ameritech, or Southwestern Bell

SNET (Southern New England Telephone)

01YYYZZZZ00

or

01YYYZZZZ000

or

XXXYYYZZZZ0101

Southwestern Bell

XXXYYYZZZZ0101

or

XXXYYYZZZZ

or

01YYYZZZZ000

or

01YYYZZZZ00

or

XXXYYYZZZZ01

Sprint/Centel

XXXYYYZZZZ100

or

XXXYYYZZZZ0000

or

SPID1 XXXYYYZZZZ000

SPID2 XXXYYYZZZZ100

Standard Telephone (GA)

SPID1 XXXYYYABCDCD

SPID2 XXXYYYEFGHGH (last two digits of phone number repeated)

US West

XXXYYYZZZZ1111

or

01YYYZZZZ00

or

01YYYZZZZ000

Verizon- See Bell Atlantic, or GTE

Vermont Telephone

XXXYYYZZZZ0101

A2 Telephone Terminology Guide

You'll get better results from the Telco if you understand and speak the lingo! We have tried to include the typical acronyms used by Telco personnel. We've put the definition under the most commonly used acronyms.

AMI – Alternate Mark Inversion. A T1 line coding method. This is the older of the two commonly available. Modified AMI is also used on the ISDN S interace. See line coding, T1. See Also B8ZS.

ANI – Automatic Number Identification- A system, originally designed for use by Interexchange carriers (IECs), that transmits the "billed party number" along with a call. Note that the billed party number is not necessarily the number of the line placing the call. ANI predates SS7 and can operate with analog as well as digital trunks. See also CLID and Caller ID.

Asynchronous Data - A form of serial data communication that is not clocked. To keep the bit stream synchronized, start and stop bits are added, which cuts down on throughput. RS-232 computer data is commonly asynchronous data. In contrast to synchronous data.

Acoustic Echo Canceller (AEC)- When an acoustic path exists from a microphone to a loudspeaker in a teleconferencing application, the far-end participant will hear his voice coming back after a delay. This echo is annoying and can disrupt conversation. An AEC is a signal processing function that works to remove this echo.

B Channel - Bearer Channel. One of the multiple user channels on an ISDN circuit. Used to carry user's data; i.e. coded audio data in the case of Zephyr or ZephyrExpress.

B8ZS- Bipolar 8 (with) Zero Substitution. A T1 line coding method. This is the more modern line coding method of the two commonly available. See Line Coding, T1. See also AMI

Bearer Channel- See B Channel

Behind the PBX- This is our own term, and refers to when one privately owned phone system is tied to another privately owned phone system. The most common application is when a key system is connected to analog ports of a PBX. When it involves one PBX behind another, it is a limited Tandem application. See Tandem Switch and Tandem Tie Trunk Switching below.

Bell Labs. The basic research facility that was AT&T's primary research facility. Bell Labs was originally part of Western Electric and was spun off with Lucent Technologies when AT&T was broken up in 1984. Many very important discoveries were made at Bell Labs including the transistor, communications theory, and radio astronomy. The future of Bell Labs seems bleak at the time of this writing.

Bell Operating Company. See BOC. See also RBOC.

Bellcore- BELL COmmunications REsearch. See Telcordia. The research and development organization owned by the RBOCs. Bellcore represents the RBOCs in developing standards for Telco equipment and in testing equipment compliance to those standards. Bellcore also offers educational and training programs open to all interested parties. Now Telcordia.

BERT - 1) Bit Error Rate Test- A test for digital lines which involves looping a data path and sending a test pattern. Data returning is compared to the sent data to check for errors. Depending on the "Test Pattern" used, BERTs may or may not uncover problems. A line, which only has occasional problems, will need a BERT of sufficient time duration to catch that intermittent problem. A five minute BERT of an ISDN BRI circuit will only catch severe problems. 2) Bit Error Rate Tester. The test equipment used to perform a Bit Error Rate Test.

Billing Telephone Number- The main phone number which all calls on hunt group or a PRI are billed to. This information may be required when configuring a PRI PBX.

Bit Error Rate- The basic measure of errors on digital transmission paths. It is usually expressed as the number of errors per number of bits. For example, the allowable bit error rate on a BRI circuit is 1 x 10 E7 (one bit error in 107 bits or 1 error in 10 million bits).

Bit Error Rate Test-See BERT

Bit Rate- The capacity of a digital channel. ISDN calls are set up at a given bit rate, either 64Kbps or 56Kbps. The bit rate of a circuit-switched connection cannot be changed during a call. See Kbps.

BLEC- Building Local Exchange Carrier. A LEC who covers the occupants of a single building (or a small group of buildings) only. Often Telecom services are provided by a BLEC as a service

or incentive to potential tenants. If a BLEC offers Long Distance Service it is covered by the same regulations as any other LEC.

Blocking- When a circuit switched call cannot be completed. The percentage of blocked calls to the number of calls attempted forms the basis of a statistic called "grade of service". While it is economically infeasible to build a network that would have no blocking, the Telcos are held accountable by the utility commissions to keep blocking below tariffed levels. The concept of blocking cannot be applied to packet networks (they just lose packets, instead), only circuit switched networks.

Blue Alarm- Also called an Alarm Indicating Signal (AIS). A keep-alive signal sent if a problem occurs mid-span in a T-carrier system. The blue alarm signal is required because in some cases T-1 repeaters will become unstable if inadequate 1's density is not maintained.

BOC - Bell Operating Company. One of the regional telephone companies that were owned by AT&T before divestiture in 1984 (i.e. New England Telephone, Ohio Bell, etc). The 22 BOCs were divided among the RBOCs at divestiture. See RBOC.

Both Way Trunks- see combination trunks.

BRI- ISDN Basic Rate Interface- The common form of ISDN with 2 "Bearer" Channels and one "D" channel. All three channels are on a single copper pair and encoded with type 2B1Q coding.

BRITE- Basic Rate Interface Transmission Extension. A technology where ordinary T-1 trunks (or any other digital carrier system) are used to extend ISDN BRI service. BRITE uses 3 DS0 channel per BRI - One for each B channel and one for the D channel. See Repeater.

BTN- See Billing Telephone Number.

Business Office- The part of the phone company where you call if they mess up your bill, to report problems, and to order service. Not necessarily technically literate.

Called Party Address- This is the destination phone number of a call delivered to a switch. For instance this could be the CLID of a call delivered to a PBX using DID or two-way trunks. See also DID.

Caller ID- A CLASS feature on an analog line that provides the number of the calling line as a burst of FSK data (modem tones) following the first ring. Also called Calling Line Identification. See also CLASS.

Calling Line ID- See CLID. See also Caller ID.

Calling Party Control- See CPC.

CAS -Channel Associated Signaling. A bit-based signaling method used on digital lines (such as T1) that is periodically inserted into the low order bit also used for the audio transmission. See Robbed Bit Signaling.

Cause Code- A code returned by switching equipment to ISDN equipment as part of the call control signaling protocol. The cause code indicates that a network call-related event has occurred or has failed. Since these codes actually come from the network, the fact you are getting a cause code is an indication that that the ISDN circuit is operational at some level.

CCIS- Common Channel Interoffice Signaling. A signaling system where network information such as address and routing information are handled externally to the actual communications (voice) path. SS7 (Signaling System 7) is the internationally standardized CCIS system. Deployment of CCIS increased efficiency since no communications (voice) channels are used merely to report an "all trunks busy" or "far end busy" conditions. It also decreased toll fraud substantially since it removed the potential for access to the signaling information that was inherent to in-band signaling schemes. CCIS also enables CLASS features as well as sophisticated re-routing features for "intelligent network" applications. See also in-band signaling. See also SS7.

Central Office- See CO

Centrex- Central Exchange Service. An enhanced business telephone service intended to offer most of the features of a PBX but where the lines are all from the LEC out of a public switch. Offers CLASS-like features for business users such as 4-digit "inside" dialing, hold, transfer, attendant, etc.

CEPT- Conference on European Posts & Telecommunications. This is a European standards body that formerly set the standards for telephone interfaces for 26 countries.

CEPT Format- The usual rate and frame format for E1 circuits. 2.048 mbps. See E1.

CEPT Rate- See CEPT format. See also E1.

Channel- An actual path you can talk or send data over. This is what you are paying the phone company for. For instance, ISDN BRI lines can be ordered with 1 or 2 active channels and these channels can be configured for voice calls (CSV), data calls (CSD) or both (alternate CSD/CSV). A channel does not necessarily have it's own unique telephone number. See ISDN.

Channel Associated Signaling- See CAS. See also Robbed Bit Signaling.

Choke Exchange- A telephone exchange, which is assigned to Radio and TV stations, Promoters, and other users that will be receiving large numbers of simultaneous calls. The idea is to group all of these users on a single exchange so when all routes into that exchange are in use "normal" users (on other exchanges) will not experience blocking of incoming or outgoing calls. Trunks from other local exchanges into the choke exchange are deliberately limited to just a few paths so callers will get an "all trunks busy" instead of completely blocking their local exchange. However, when one of the choke exchange users experiences a large number of calls (as when

your station runs a contest) the other choke exchange users will be blocked because all trunks into the choke exchange will be busy.

In the modern network, using CCIS signaling such as SS7, actual trunks are not used to convey "busy" or "all trunks busy" conditions. Thus blocking due to a station contest should not occur as the busy status in response to a call attempt is conveyed over the separate SS7 network. Therefore, the need for choke exchanges has pretty much disappeared. Nonetheless, many Telcos still insist that Broadcasters use special choke lines for call-in lines.

See blocking and concentration.

Circuit- A physical path through which electrical signals can pass. It consists of a network of conductors and other components, separated by insulators. Technically this term cannot be applied to fiber optic or other "non-metallic" paths. See also channel.

Circuit Switched Data- See CSD

Circuit Switching- A system where a dedicated channel is allocated to the users of that call for the duration of that call. That channel is allocated for the duration of the call regardless if information is being transmitted at any given moment. Bandwidth through the channel is fixed, at no time may this bandwidth be exceeded. If this bandwidth is not used it is wasted. While inherently inefficient, the dependable and reliable nature of circuit switching makes it ideally suited to real-time voice and audio/video conferencing applications. When over loaded Circuit Switched networks will respond "all circuits are busy… try again later". This is in stark contrast packet switched networks or to systems where statistical multiplexing is used. See Statistical Multiplexing.

CLASS- Custom Local Area Signaling Services. A variety of enhanced features (usually on analog lines) that take advantage of the ability of modern SS7 technology's ability to transmit information about the calling party. CLASS includes such features as Caller ID, Automatic Callback, Call Trace (initiated by subscriber), Selective Call Screening, etc.

Clear Cause. See Cause Code

CLEC- Competitive Local Exchange Carrier. Your local telephone service provider who is one of the new-generation providers rather than a RBOC or Independent. A CLEC is really just an independent, albeit one formed after the divestiture of AT&T. See LEC and Independent.

CLI - Calling Line Identity. European term for CLID. See CLID.

CLID- Calling Line Identification. This is the ISDN and SS7 equivalent of Caller ID; I.E. the number of the calling party. See also Caller ID and ANI.

CO- Central Office. The Telco facility to which your local telephone circuit lead. Contains "Switches" and "Trunks" as well as the local telephone circuits.

Codec- COder/DECoder. A device which takes digitized audio and "codes" it in order to reduce the transmission bit rate and which can also simultaneously "decode" such coded audio. Strictly speaking, a codec does not include an ISDN terminal adapter and related equipment.

Simple codecs are also used in digital telephony. These use a simple companding scheme to reduce channel noise.

COL - COnnected Line number. European Term. The number to which you have connected. This may not be the number you dialed if call forwarding is used.

Combination Trunk- A trunk (channel) which can both make and receive calls. This generally refers to analog ground start or loop start trunks, although the term can be applied to ISDN BRI or PRI channels as well. Each combination trunk normally has a telephone number, although they are frequently part of a hunt group and only one number may be published for that group. Also called a Both Way Trunk. This is not the same as a Two-way DID trunk. See DID trunk, Hunt Group and Trunk.

Common Channel Interoffice Signaling- See CCIS

Competitive Local Exchange Carrier- See CLEC.

Concentration- The basic premise is to share facilities wherever possible. For instance, while there may be thousands of customers served by a given Central Office, there will be substantially less than that number of calls which can be handled simultaneously. And, even fewer long distance calls can be made simultaneously. The art of Traffic Engineering is to have enough capacity that calls are rarely blocked, but no more than that number. See also Choke Exchange and Blocking.

CPC- Calling Party Control. Sometimes referred to as "CPC Wink" or "disconnect supervision". A call supervision feature on an analog loop start line that provides the ability for a CO (Central Office) to signal the called party when the calling party hangs up. CPC allows the PBX, key system, or telephone answering device to reset the line so that it is ready to accept or initiate another call. CPC is accomplished by either a loop current drop or reversal. With some CO equipment, it is also provided if the called party drops the call. See also MCLD.

CPE- Customer Premise Equipment- Customer owned equipment located at his/her facility, such as a CSU or terminal. In the USA and Canada, the NT1 is part of the CPE.

CPN - Called Party Number - European Term. The number that has been dialed. See Called Party Address.

CSD- Circuit Switched Data- A dial-up data communications channel which, once established, looks like a transparent data pipe. Also, the type of ISDN service required to utilize this capability of an ISDN circuit. In contrast to CSV.

CSU- Channel Service Unit. The NCTE used in the USA & Canada to terminate a T1 line. Typically the CSU must be provided

by the end user. See NCTE and DSX1

CSU/DSU- A device which incorporates the functions of a CSU (Channel Service Unit) and a DSU (Data Service Unit) Most commonly it interfaces between a Switched-56 or Dedicated Digital Service circuit and a user's data equipment such as the Zephyr.

CSV- Circuit Switched Voice- A dial-up communications circuit for voice grade communication. Also, the type of ISDN service required to use this capability of an ISDN circuit. In contrast to CSD

Custom ISDN (USA & Canada)- An ISDN protocol which pre-dates National ISDN-1. In most cases National ISDN-1 is also available. The Northern Telecom DMS-100 switch Supports "DMS Custom Functional" ISDN. The AT&T/Lucent 5ESS switch Supports "Custom Point-to-Point" (PTP) and Custom Point-to-MultiPoint (PMP). The ISDN protocol has no relation to where one may call. Telos equipment does not support PMP.

D Channel- Data/Delta Channel. Depending on who you ask, it is Data or Delta. The channel which handles ISDN network-related data between the user's equipment and the Telco switch. Used to carry data to set up calls and receive calls. Some Telco's also allow users to use the D channel to access the packet data network, with appropriate terminal equipment.

D4- See Superframe. See also Line Format.

DCE- Data Communication Equipment. When using serial communications such RS-232, V.35, or X.21, the DCE is the device sending/receiving from the Telco line. i.e.: a modem or CSU/DSU. In contrast to DTE.

DDS- Digital Data System- See Dedicated Digital Service.

Dedicated Circuit- A permanent channel between two locations. As opposed to a Switched Circuit.

Dedicated Digital Service- a "Hardwired" or "Nailed Up" digital circuit which is permanently connected between 2 points. Typically 56Kbps or 64Kbps. Dedicated digital lines are frequently cheaper than ISDN for full time service. Also called Digital Data System or DDS.

DID- Direct Inward Dialing. The ability for an outside caller to dial to a PBX extension without going through an attendant or auto-attendant. See also DID Number and DID Trunk.

DID Extension or DID station- A specific phone within a PBX which can be called from the public telephone network without going through an attendant or auto-attendant.

DID Number- A phone number used to route calls from the telephone network to a specific phone in a PBX (the DID extension). DID requires special DID trunks or ISDN PRI "two-way DID" trunks. Blocks of DID numbers (typically 10 or 20, sometimes higher) are purchased from the LEC or CLEC for use on the PBX. The number of DID numbers usually substantially exceeds the number of trunks in the system.

DID Trunk- A Direct Inward Dialing Trunk. A trunk (channel) which can only receive calls. A group of telephone numbers (DID numbers) are associated with a given trunk group, however there is no one-to-one correspondence between the individual channels and these numbers. The PBX uses the DID number given it by the phone company to route the channel to the correct DID extension within the PBX. This allows some or all PBX stations to receive calls directly without going through an attendant (or auto attendant) Note that there are usually more DID numbers than there are DID trunks. See DID number and DID extension.

Direct Inward Dialing- See DID

Directory Number (USA & Canada)- Your seven digit telephone number (without the area code), as found in the telephone directory.

Directory 1&2 (Zephyr)- The Utility menu on the Zephyr where the 7 digit Directory Numbers can be entered during set up. The Zephyr generally does not require these numbers.

DN- See Directory Number. See also Directory 1 &2 and MSN/DN 1 &2.

DNIS- Dialed Number Identification Service- A service, typically offered by a long distance company on 800 lines, that provides the number dialed by the caller. This allows a caller to receive specific treatment depending on the number dialed.

DP - Dial Pulse. A method off sending address information by either 1) Causing brief interruptions in loop current. or 2) Causing brief changes of state of a bit on a digital circuit using Channel Associated Signaling. In other words, "rotary" or "pulse" dialing. See also DTMF and CAS.

DSØ- Digital Signal Level Zero. The smallest unit of measure of the standard rate hierarchy used by the Telcos (i.e. all other rates are a multiple of the DSØ rate. For example, the T1 rate is 24 times the DSØ rate and the E1 rate is 32 times the DSØ rate). 64 kbps. See also B channel.

DS1-Digital Signal Level 1. The second level up the digital rate hierarchy used by the Telcos. This is 24 times the DSØ rate for a total of 1.544 mbps. See DSØ. See also T1.

DS2-Digital Signal Level 2. Data rate of 6.312 mbps (4 times the DS1 rate). See DSØ and DS1.

DS3-Digital Signal Level 3. Data rate of 43.232 mbps (28 times the DS1 rate). See DSØ and DS1.

DSL- Digital Subscriber Line. Traditionally refers to an ISDN circuit or sometimes a T1 line, although the term is also frequently used to mean the next generation beyond ISDN. Sometimes xDSL is used to indicate that the writer is referring to any of a number of emerging DSL technologies.

DSU- Data Service Unit. See CSU/DSU.

DSX-1- Digital Cross Connect level 1 (USA & Canada, primarily). Defined as part of the DS1 (T1) specification and is a closely related signal. The type of signal switched by a Digital Cross-Connect System (DACS). The FDL is stripped off at the DACS interface. DSX-1 is also the type of signal that arrives at the user side of a CSU on a T1 line. A DSX-1 cable is limited to 655 feet (200 meters).

DTE- Data Terminal Equipment- When using serial communications such RS-232, V.35, or X.21, the DTE is the device sending/receiving from a modem or CSU/DSU. In contrast to DCE.

DTMF – Dual Tone Multi Frequency. The standard tone-pairs used on telephone terminals for dialing using in-band signaling. The standards define 16 tone-pairs (0-9, #, * and A-F) although most terminals support only 12 of them (0-9, * and #). These are also sometimes referred to as "Touch Tones". Note that while digital data terminals have the same symbols, ISDN uses "common channel signaling" (over the D channel) and therefore does not necessarily generate any tones at all. However many terminals still generate the tones since they will still be used on occasion to access services (such as voicemail or automated attendant) at the far end using in-band tones.

E1- A common type of digital telephone trunk widely deployed outside the US and Canada. Has 31 available 64 kbps channels (called DS0s) plus a sync/control channel for a total data rate of 2.048 mbps.

E-1- See E1

ESF- Extended Superframe. A type of Line format supported on T1 circuits. The Telco determines the line format and line encoding of your line. See Line Format.

ETS 300- The pan-European ISDN protocol standardized by ETSI. This protocol is used throughout Europe and has been adopted in many other countries outside the USA & Canada. See also MSN.

ETSI - European Telecommunications Standards Institute

Euro ISDN- See ETS 300.

Exchange- Another name for a Central Office (most often used in European countries). Also used in the USA & Canada to refer to a particular 3-digit prefix of a 7-digit telephone number. See CO.

Extended Superframe. See ESF.

FDL- Facilities Data Link. A bi-directional data link available on T1 circuits when the ESF line format is used. The FDL is primarily used by the Telco to poll the CSU for error statistics.

Four Wire – A circuit path using separate pairs for send and receive. This term is also used when referring to digital channels that inherently have discrete send and receive paths, regardless of the number of pairs (or other media) used. See also Hybrid.

Frame – A unit of data which is defined by the specific communications protocol used. See Line Format, T1.

FXO – Foreign Exchange Office termination. A line or port meant to connect to connect to the POTS output of a Central Office. Such an interface goes off and on hook to signal status and expects to receive ringing current.

FXS – Foreign Exchange Station termination. A line or port mean to connect to a telephone. Such an interface must look for current flow to know when the attached device goes off-hook. It must be able to generate ringing current.

Glare – On a POTS line an incoming call is signaled by periodically applying an AC ring voltage to the line. Since there is a semi random period before the ring, and pauses between rings, it is possible to seize a line which is "about to ring" (and answer a call) when attempting to place an outgoing call. When this scenario happens it is called glare. Glare is much less likely if Ground Start or ISDN trunks are used. See also Ground Start Trunk.

GR-303 - See SLC-96

Grade of service- This is simply the ratio of calls blocked to total calls in a decimal form. Therefore, a grade of service of P.08 would represent 8% blocking. Telephone tariffs regulate the acceptable average grade of service which must be provided on public networks. See also Blocking.

Ground Start Trunk – A type of telephone trunk where the request to make an outgoing call (i.e. request for dial tone) is made by briefly grounding the Tip conductor. Many PBX system use ground start trunks as they are less prone to glare than Loop start trunks. Ground start lines are sometimes used with equipment designed for Loop Start lines. This may or may not work - generally it serves to prevent outgoing calls while incoming calls work normally. Telcos may call these "ground start lines". See Loop Start Trunk. See also Glare.

HDB3- High Density Bipolar 3. An E1 line coding method. This is the more modern line coding method of the two commonly available. See Line Coding, T1. See also AMI and B8ZS

Hunt group- A group of telephone channels configured so that if the first is busy (engaged) the call goes to the next channel, if that channel is busy it goes to the next channel, etc. Hunt groups may hunt from the highest to the lowest, the lowest to the highest, or on some other arbitrary pattern. But the order of hunting will usually be fixed, beginning with one channel and working through ("hunting") until an unused channel is found. The term may have originated back in the old manual switchboard days when the operator literally hunted for an unused jack to plug a cord into. This arrangement is very common in business scenarios where a single incoming number (the Listed Directory Number) is given to the public, but multiple incoming channels are supported. See also LDN.

Hybrid – A device which converts from a two-wire signal such as

POTS lines (or a 2-wire intercom) to a four-wire system (separate send and receive paths) such as used in the pro-audio world. While this task is theoretically quite simple, the fact the impedance of most phone lines varies widely across frequency complicates matters. The Telos 10 telephone system was the first practical DSP based hybrid and applied the then brand-new technology to this problem.

IEC - 1) InterExchange Carrier. "Long Distance" carrier. Handles Interlata and interstate calls. Also referred to as IXC. 2) International Electrotechnical Committee. A European standards body best known for the power plug now used throughout the world for AC power cords for use on office equipment and computers.

ILEC – Incumbent Local Exchange Carrier. A local Exchange Carrier which entered the marketplace before the enactment of the 1996 Telecom act;. i.e. a telephone company which is neither an Indi nor an RBOC. See LEC and CLEC.

IMUX – See Inverse Multiplexing

In Band Signaling- A signaling system where network information such as address and routing information are handled over the communications (voice) path itself. Usually the information is represented in the form of audible tones, however DC loop current signaling also qualifies as In Band Signaling. See also CCIS.

Incumbent Local Exchange Carrier. See ILEC. See also CLEC & LEC

Independent – Any of the phone companies in existence at the time of divestiture that were not affiliated with the Bell System. See RBOC, LEC, and CLEC.

Indi- See Independent.

Interconnect Company- A vendor of telecommunications CPE other than a BOC or AT&T. This term was originated by AT&T and was meant to be derisive towards the fledgling industry when the courts said it was OK for end users to buy equipment from someone other than the Bell System. This industry flourished, in spite of AT&T's disdain, and ironically, the RBOCs were not allowed to sell CPE under the terms of the break up of AT&T. With the current state of deregulation, the RBOCs are slowly re-entering this business.

Interexchange Carrier- See IEC

Interwork- The ability of two different type of networks to communicate seamlessly. For example, ISDN can interwork calls to both the POTS network and the Switched-56 network.

Inverse multiplexing- A method of breaking apart and later combining 2 data streams that does not involve the telephone network. Inverse multiplexing is the most common way of achieving this function in Codecs operating at 112 or 128kbps over ISDN.

IOC Capability Packages - ISDN Ordering Code system. This system was devised by the National ISDN User's Forum and Bellcore to simplify ordering new ISDN lines in the USA and Canada. Using a single code specifies all line specifications. The Telos Zephyr, ZephyrExpress, TWO, and TWOx12 support IOC package "S". Search for document NIUF 428-94 at NIUF's web page http://www.niuf.nist.gov/ for more information.

ISDN - Integrated Services Digital Network- A relatively new and highly flexible type of telephone service which allows dialing on digital channels with multiple bi-directional "Bearer" channels each with a capacity of 56 or 64 Kbps and a single bi-directional "D channel". See BRI and PRI.

ISDN Protocol - The "language" used for communication between the Telco's switch and the customer's Terminal Adapter. Each ISDN circuit has one protocol, and the protocol has no effect on where or whom one may call. See ETS 300, National ISDN, and Custom ISDN.

ISDN 2- A term used in Europe for ISDN BRI. Also called S . Not to be confused with National ISDN-2. See BRI.

ISDN 30- A term used in Europe for ISDN PRI. Also called S2M See PRI.

ISG – Incoming Service Grouping. A Hunt Group. See Hunt Group

IXC- IntereXchange Carrier- See IEC

kbps- KiloBits Per Second. Measure of digital channel capacity.

Key Telephone System – A system that allows multiple telephones to share multiple pre-determined telephone lines. The system provides indicators to allow the user's to understand the status of each line available on a given phone. In its most basic form it is up to the user to provide the intelligence to select an unused line, or answer a ringing line, for example. The Telos 10, Direct Interface Module, One-x-Six, TWOx12, Nx12, and Series 2101can be classified as Key Telephone Systems. See also PBX.

LATA- Local Access and Transport Area. The area within which calls are routed by your Local Exchange Carrier (LEC). Under the divestiture of the Bell System calls going outside of this area must be handled by an Interexchange carrier (IEC). With the latest round of de-regulation the usual IEC companies are being allowed to compete in the IntraLATA long distance market and LECs are beginning to be permitted to handle InterLATA calls.

LDN- Listed Directory Number. When a number of Telco channels share the same hunt group, it is customary to give out only one phone number for the group, although generally each channel will have its own number. The number given out is the "Listed Directory Number" since that is the number that would be listed in the Telephone Directory and given to customers. Sometimes called a Pilot Number. See also DN and Hunt Group.

LE - Local Exchange. European term for Central Office. See CO.

LEC- Local Exchange Carrier. Your local telephone service provider which is either an RBOC or an Independent. In other words, a traditional phone company. In contrast to CLEC or IEC.

Line- An electrical connection between a telephone service provider's switch (LEC or CLEC) and a telephone terminal or Key system. An electrical connection between a telephone service provider's switch and another switch is technically called a trunk. Note that some type of physical lines offer more than one channel. I.E. a BRI circuit has 2 channels, called B channels. This term is a confusing one, so we try to avoid using it. See Channel. See also Station Line.

Line card- The circuit in the Telco switch to which your line is connected. On an ISDN circuit the line card performs a role analogous to the NT1 in adapting to and equalizing the circuit to establish OSI Layer 1.

Line Coding, T1- The clock signal for T1 is derived at the far end from the data bits themselves. Therefore, T1 lines have certain restrictions as to the data allowed. No more than 15 zeros shall be sent in a row; and average density of 12.5% ones must be maintained. The CSU is responsible to ensure that these requirements are met. The line encoding method, AMI or B8ZS determines exactly how these requirements are met while still allowing recovery of the original data at the far end. Your Telco will determine the method used on a specific circuit. B8ZS is preferred. E1 circuits have similar restrictions. HDB3 is preferred for E1 circuits.

Listed Directory Number- See LDN

Line Echo Canceller (LEC)- When a two-way digital path is interfaced to an analog line, echoes result from the leakage of the send audio into the receive. When combined with delay, these are annoying and can disrupt conversation. An LEC works to remove these echoes, usually using an adaptive digital filter. See Hybrid.

Line Encoding, T1- See Line Coding, T1.

Line Format, T1-Modern T1 circuits usually use either Superframe (sometimes called SF or D4) or Extended Superframe (sometimes called ESF) line formatting. The type of framing used is determined by your Telco. ESF is preferred. See ESF and SF

Line Side- This is the side of a central office switch that the subscriber's telephone lines are connected to. The main reason for distinguishing between this and the trunk side is that certain customer related features (Such as CLASS and Centrex features) are inapplicable to trunks. See also Trunk Side. 2) The user side of a PBX. Also called the station side.

Line Termination - See LT.

Local Access and Transport Area- See LATA.

Local Exchange Carrier- See LEC and CLEC.

Long Distance- If your local Telco is a former Bell Operating Company then any call outside of your LATA or any Interstate call is considered long distance and is handled by an IEC. The above is true regardless of whether you are referring to a dedicated line or a dial up call. 2) However, under the current state of deregulation, toll calls within a LATA may now be covered by the IEC, and in some cases RBOCs are being permitted to handle InterLATA calls.

Loop- The telephone circuit from the CO to the customers premises. Generally refers to a copper cable circuit.

Loop Current Disconnect Supervision - Another name for CPC. See CPC

Loop Qualification- Process of actually measuring the loss on a prospective ISDN line to see if it can be used for ISDN service. The actual loss on the line (usually measured at 40 kHz) is the determining factor whether ISDN service can be offered without a repeater. Generally ISDN is available up to 18,000 feet from the serving Central Office. It may not be available within this range, or may be available further from the CO. Only a loop qualification can tell for sure. Not all Telcos will extend ISDN lines with repeaters.

Loop Start Line - A plain old telephone line. The telephone terminal signals the "off hook" condition by allowing DC current to flow. See Ground Start Trunk. See also Glare.

Loop Start Trunk – A plain old telephone line connected to a PBX switch. See Loop Start Line. The PBX signals the "off hook" condition by allowing DC current to flow. Ground Start Trunks are generally preferred for use on PBXs to prevent glare. See Ground Start Trunk. See also Glare. The Telco may call this a "Loop Start Line".

LOS- Loss Of Signal. An LED or other indicator that illuminates if a signal is absent. This terminology is commonly used with T-1 equipment.

LT - Line Termination - The electrical and protocol specifications for the Central Office end of an ISDN line. If you wish to connect an ISDN terminal (such as a Zephyr Xstream) to a PBX the PBX must support LT ISDN. See also NT and Line Card

Lucent Technologies- Company which now makes the former AT&T 5ESS switch, as well as various other piece of Telco gear and semiconductors. Lucent was split off from AT&T in 1996. As of approximately 1999 Lucent sold their telephone set manufacturing business (and the right to use the AT&T name on telephone sets) to V-Tech. V-Tech is using the AT&T name and line as their high-end line. Lucent's PBX division was spun off in 2001 as AVAYA.

MCLD- Modifying Calling Line Disconnect. The parameter on the Lucent 5ESS switch that determines if CPC is active. Should be set to "Yes" if CPC is required. See CPC

MOS Grade- "Mean Opinion Score" Grade is a measure of codec quality, ranging from 1 to 5. These are derived from subjective tests with listening panels comprised of both experts and "people from

the street" who compare the coded audio to the original and grade the relative quality.

MSN/DN 1&2 (Zephyr Xstream)- The Telco menu selections on the Xstream where the MSN (Euro & INS 64 ISDN) or 7 digit Directory Numbers (USA ISDN) can be entered during set up. The Zephyr Xstream generally does not require that these numbers be present. See Directory Number and MSN.

MSN- Multiple Subscriber Number (Euro ISDN). This is a telephone number associated with an ETS 300 BRI line. Providers of ETS 300 often give you three MSNs with a BRI, although additional MSNs can be purchased. An ISDN terminal will "ring" (provide an alerting signal) only when calls are made to the MSN (or MSNs) entered in that terminal. If a terminal has no MSNs entered, it will "ring" whenever there is a call to any of the MSN's on that BRI. See ETS 300 and DN.

National ISDN (USA & Canada)- The US "standardized" multi-platform ISDN protocol. The first version is National ISDN-1. As of mid 1996 National ISDN-2 has been implemented in some areas and is fully backward compatible with National ISDN-1.

NCTE – Network Channel Terminating Equipment. NCTE is a general term that can be applied to a CSU or NT1 or other equipment terminating a digital line at the customer's premises. In many countries, the NCTE is provided by the Telco. The USA is not one of those countries.

Network Channel Terminating Equipment. See NCTE.

Network Termination - See NT

Nortel- Manufacturer of the DMS-100 switch as well as many other pieces of telecom and network equipment. Now called Nortel Networks.

NT - Network Termination - The electrical and protocol specifications for the user end of an ISDN line. See also LT.

NT-1- An alternative expression for NT1. See NT1

NT1- Network Termination Type 1. The termination at the customer premises of an ISDN BRI circuit. The NT1 performs the role of line termination of the "U" interface and Codes/Decodes from the line's 2B1Q coding scheme. The customer end of the NT1 interfaces using the "S" or "T" interface. The NT1 is frequently part of the "Terminal Adapter" and is built-in to Zephyr Xstream, Zephyr, ZephyrExpress, Telos TWO and TWOx12 systems sold in the USA & Canada. See also NCTE

NTBA- Network Termination Basic Access. The term used for NT1 in some countries. See NT1. See also NCTE.

OOS – Out of Service. An alarm light or condition on a T1 or trunk.

PBX- Private Branch Exchange. A privately owned switch. Basically, a PBX is a private "business" telephone system which also interfaces to the telephone network. In some circles 'PBX' implies a manual switchboard whereas 'PABX' (Private Automatic branch exchange) implies a PBX that supports dialing by end users.

Many PBX's can now offer ISDN BRI service, usually over the S Interface. A few vendors are now offering BRI over the U interface as well. PRI over DSX-1/T1 or E1 is also offered in some cases. Be wary of these ISDN protocols since they have not been as well tested as the versions running on "public" switches. They may or may not work with a given piece of CPE.

PIC- Primary Interexchange Carrier. (USA) This is your default "1+" carrier used for interLATA calls. In some areas you may have two PICs, one for interLATA calls, and one for intraLATA long distance calls (in which case it stands for Primary Intraexchange Carrier). In some areas intraLATA long distance calls are still handled by your RBOC, in others you now have a choice. You may be able to discover who you current PIC is by dialing 700 555-4141.

PMP (USA) - AT&T "Custom Point to Multi-Point" Custom ISDN Protocol. Not supported by the Zephyr or ZephyrExpress. See Custom ISDN and ISDN Protocol.

POP - Point Of Presence. The local facility where your IEC maintains a switch. This is where your long distance calls get routed so that your IEC can handle them. Also used to describe the local access point of an Internet Service Provider.

POT- Plain Old Telephone. A black, rotary-dial desk phone. Usually a Western Electric model 500 set. Outdated term.

POTS – Plain Old Telephone Service. Regular old-fashioned analog loop start phone service.

PRI – ISDN Primary Rate Interface- A form of ISDN with 23 "B Channels" and one "D channel". All 24 channels are on a single cable. Functionally related to T1 telephone circuits. In Europe PRI has 30 "B Channels" and one "D Channel" and one "Sync channel". See also B channel and D channel.

Provisioning -The act of configuring an ISDN or other telecommunications path. Also refers to the complete line configuration information.

PS2 Power - Power provided on pins 7 and 8 of the "S" interface cable. This power is used so that a NT1 can provide power to a terminal (usually a phone). In some cases, it is used to allow a terminal to power an NT1. The USA versions of the Zephyr and ZephyrExpress supply PS2 power in the "S" jack. The Zephyr Xstream and Telos TWO provide this on all versions. This power arrangement is also used in the Telos TWOx12 and 2101 Studio Interface to power Desktop Directors.

PTP (USA)- AT&T "Custom Point to Point" Custom ISDN Protocol. Point-to-Point lines have only one incoming phone number which must be dialed twice to connect to both lines (the first call goes to "line 1" and the second call rolls over to "Line 2". See Cus-

tom ISDN and ISDN Protocol.

RBOC- Regional Bell Operating Company. (USA) One of the regional companies formed when AT&T was forced out of the local telephone business. Each RBOC (or "baby bell") owns a number of the former "Bell Operating Companies". The Bell Operating Companies are the traditional local phone companies (pre-1984), except where one's service is from an "Independent" (non bell) telephone company or a CLEC. Due to their former association with the Bell System RBOCs are regulated by the FCC differently than are independent Telcos or CLECs. In many cases the Bell Operating Company structure is no longer used. For instance, here in Ohio we now deal directly with the RBOC, Ameritech, while the old Bell Operating Company, Ohio Bell Telephone, no longer exists. Another trend is mergers among the RBOCs (and in some cases the independents as well). See CLEC and LEC.

RD- Receive Data. Data coming from the network, or DCE towards the DTE. Also, a light on a modem or CSU/DSU that lights to indicate presence of this signal.

Red Alarm- An alarm state on a T-carrier circuit that indicates that the incoming signal (at the network interface) has lost frame for more than a few seconds. Normally a Yellow alarm is then returned (i.e. sent back) if a Red alarm is present. A Red Alarm indicates a loss of inbound signal; a Yellow alarm indicates (indirectly) a loss of outbound signal. See also Yellow alarm, Blue alarm, and LOS.

Regional Bell Operating Company- See RBOC

Repeater- A device intended to extend ISDN telephone service to sites further from the central office than could normally be served. i.e.: beyond 18,000 feet. ISDN repeater technologies include "BRITE", "Virtual ISDN", "Lightspan", and "Total Reach". Some Telcos do not use repeaters. Compatibility between a given NT1 (CPE) and a repeater is less certain than if that CPE where directly connected to the switch.

Robbed Bit Signaling- A signaling scheme that "borrows" bits on each T1 channel for use as signaling channels. On SF T1's there are two bits, the A bit and the B bit in each direction. On ESF T1's there is also a C and D bit in each direction, although they are rarely used. Using these bits, various older analog trunk interfaces can be emulated over a T1. For instance, dial pulse address signaling using 10 pulse per second (rotary style) digit groups over these bits. Since robbed bit signaling interferes with the least significant bit, only 7 bits can be used for sensitive data applications, leaving only a 56kbps channel for data applications. See also CAS and CCIS.

Rollover – See Hunt Group

S interface- The electrical interface between the NT1 and the Terminal Adapter or other ISDN equipment. ISDN equipment with built-in NT1's do not necessarily provide access to the S interface (the Zephyr Xstream, Zephyr, ZephyrExpress and Telos TWO do). Multiple devices can share an NT1 by connecting on the S interface. Also known as the S passive bus.

S0- European term for ISDN BRI. See BRI and ISDN2.

S2M- European term for ISDN PRI. See PRI and ISDN 30

Sealing Current- Unlike telegraphy, teletypewriter and POTS lines, most digital lines (such as ISDN) use a voltage rather than current mode of operation. Sealing Current allows a controlled amount of current to be passed through a telecom circuit for purposes of "healing" resistive faults caused by corrosion. Bellcore specifies sealing current on the ISDN U interface in the USA. The Siemens EWSD switch does not provide sealing current. Most other ISDN capable switches used in North America do.

SF- Superframe. A type of Line format supported on T1 circuits. The Telco determines the line format and line encoding of your line. ESF is the preferred Line Format on T1 circuits. See Line Format.

Silence Suppression- See Statistical Multiplexing.

SLC-96 – A Subscriber Loop Carrier Circuit system manufactured by AT&T (now Lucent). SLC-96 has its own version of T1 framing between it and the CO. SLC-96 and similar "SLIC" systems may or may not perform a concentration function. The interface is the Bellcore TR-008 or the newer GR-303 interfaces that are specialized versions of T1 intended to allow transparent transport of analog CLASS features such as Caller ID, Call Waiting, etc. The GR-303 interface is specifically intended to be used as a common point of interconnection between alternative equipment, technologies, and/or networks (i.e. voice-over-DSL, voice-over-IP, etc)) and the public switched network. See the following link for additional information from Telcordia: http://www.telcordia.com/resources/genericreq/gr303/index.html

SLIC- 1) Subscriber Line Interface Circuit, see "Line Card". 2) The equipment used with the AT&T (Lucent) SLCC Subscriber Loop Carrier Circuit, a system used to multiplex a number of subscriber loops onto a single circuit (usually a T1 circuit) to reduce fixed costs. 3) Also sometimes used generically for other brands of similar equipment. See also SLC-96.

SPID- Service Profile IDentifier- On the "National ISDN", "AT&T Custom PMP" and "Custom DMS" ISDN BRI protocols, the Telco switch must receive correct SPID(s) from the CPE before it will allow access to ISDN service. Intended to allow multiple configurations on ISDN lines shared among different types of CPE equipment. While your SPID may include your area code and telephone number, the SPID is distinct from the telephone number. For the National ISDN, Custom PMP, and DMS custom ISDN protocols the equipment requires that the user to program SPIDs into it. Custom PTP and ETS 300 protocols do not require a SPID. No SPIDs are required for ISDN PRI protocols.

SS7 - Signaling System 7. The internationally adopted Common Channel Interoffice Signaling (CCIS) system. Previous to SS7 the Bell System used SS6 which did not support the International

Standards. SS7 does. It allows for substantially flexibility and power in dynamically routings calls. An SS7 database lookup is how a call to a mobile telephone user can handled transparently despite the fact that the user's location may change. Also used to determine what carrier should handle a given toll free call. See also CCIS.

Station Line – A telephone circuit from a PBX to a telephone on that PBX. Since this is a telephone-to-switch connection it is considered to be a "line". See Line and Trunk

Station Side - The user side of a PBX. The side of the switch that the telephones are attached. Also, occasionally called the 'line side'. The main reason for distinguishing between this and the trunk side is that certain customer related features (Such as Hold and Transfer) are inapplicable to trunks. See also Trunk Side.

Statistical Multiplexing- A method of improving effective bandwidth of a Telco channel. Data is divided into packets, which is sent on the channel when there is space available. This is the key to the Internet.

Subscriber- The customer of a Telecommunications company. This term dates back to when a local Telephone Company was formed at the specific request of a group of customers who agreed in advance to "subscribe" to the service.

Superframe- See SF

Switch- Telephone switching device which "makes the connection" when you place a call. Modern switches are specialized computers. ISDN service is provided from a "Digital" switch, most commonly (in the USA & Canada) an AT&T model "5ESS", Northern Telecom model "DMS-100", or Siemens model "EWSD". The switch, and related software running on it, will determine which ISDN protocol(s) will be available to customers connected to it. See also PBX.

Switched Circuit- A channel which is not permanent in nature, but is connected through a switching device of some kind. The switching device allows a switched circuit to access many other switched circuits (the usual "dial up" type of telephone channels). Once the connection is made however, the complete capacity of the channel is available for use. As opposed to a dedicated circuit or a packet based connection.

Synchronous Data- A form of serial data which uses a clock signal to synchronize the bit stream. Since, unlike asynchronous data, no start and stop bits are used, data throughput is higher than with asynchronous data. ISDN and T-1 use Synchronous data. See also Asynchronous Data.

T-Link- A proprietary Telos 2.048mbps channelized link. This link uses the DSX-1 electrical protocol and has 30 channels at 64kbps each.

T1- A common type of digital telephone carrier widely deployed within the US, Canada, and Japan. Has 24 64Kbps channels (called DS0's). The most common framing scheme for T1 "robs" bits for signaling leaving 56kbps per channel available for data use.

T-1- An alternative expression for T1. See T1.

TA- Terminal Adaptor. The electronic interface between an ISDN device and the NT1. The terminal adapter handles the dialing functions and interfaces to the user's data equipment as well as to the NT1 on the "S" or "T" interface.

Tandem Switch- A switch which is between two others. It connects two trunks together. Long distance calls on a LEC line go through a long distance tandem that passes them through to the long distance provider's switch.

Tandem Tie Trunk Switching- When a PBX switch allows a tie line call to dial out of the switch. For example, if switch "A" in Arkansas has a tie line to switch "B" in Boise, Boise could use the tie line to make calls from switch "A".

TD- Transmit Data. Data coming from the DTE towards the DCE or network. Also, a light on a modem or CSU/DSU that lights to indicate presence of this signal.

TEI - Terminal Endpoint Identifier. An OSI Layer 2 identifier used by an ISDN terminal to communicate with the serving CO. Most equipment uses a dynamic TEI assignment procedure .

Telco- Telephone Company. Your local telephone service provider. In the 21st century you generally have a choice of Telcos if you are a business in a major metropolitan area in the USA. Competition is coming to the Telecom industry around the world.

Telcordia Technologies- Formerly Bellcore. The research and development organization owned by the telephone companies. Telcordia represents the phone companies in developing standards for Telco equipment and in testing equipment compliance to those standards. Promotes competition and compatibility through standards promoting interoperability such as GR-303. Telcordia also offers educational and training programs open to all interested parties. Bellcore was sold to SAIC in 1997. Telcordia is responsive to both RBOCs and independent Telcos. Their web site is: http://www.telcordia.com.

Telos Customer Support +1 216.241.7225. Generally available from 9am to 6pm EST, but you can try at other times. You may also ask for assistance by emailing to support@telos-systems.com

Telos Test Line(Xstream, Zephyr & ZephyrExpress)- 24 Hour test line which can be called to verify correct operation of your Zephyr or ZephyrExpress codec. Numbers are +1 216-781-9310 and +1 216-781-9311. The test line transmits in dual/mono mode at 32KHz and can be received in L3 Mono using one B channel or L3 Stereo using 2 B channels. Telos also maintains an identically configured test line in Freising, Germany at 49 81 61 42 061 (dial this number twice).

Terminal Adapter- See TA

Tie Line- See Tie Trunk.

Tie trunk- A Trunk between two PBXs. Note, a tie line is a dedicated circuit, not a switched circuit. See Trunk.

TR-008 - See SLC-96

Trouble Ticket-A Telco "work order" used to track Customer Repairs within the Telco. If you call someone "inside" the Telco's repair department, they will need this number to proceed. It will also be needed whenever you call to check on the status of a repair. Always ask for this number when initiating a repair request.

Trunk- A communications path between two switching systems. Note that many trunks may be on a single circuit (if that circuit has multiple channels). The trunks most users will deal with are between the Telco switch and a PBX. However, a Tie Trunk can connect two PBXs. See also Tie Trunk and Trunk Group

Trunk Group- A number of telephone channels which are functionally related. Most common is the Hunt Group. Other common types include Incoming Trunk Groups and Outgoing Trunk Groups. See also Combination Trunks and Two-way DID Trunks.

Trunk Side- 1) The side of a central office that faces the network. Historically many CO switches could not make trunk to trunk connections (as opposed to tandem and long distance switches, that are always used to connect trunk together). Hence the need to distinguish between the "line side" and the "trunk side" of the switch. See also Line Side.

2) The side of a PBX that connects to the Telco. Historically many PBXs could not make trunk to trunk connections. Hence the need to distinguish between the "line side" and the "trunk side" of the switch. Since a trunk is a switch-to-switch circuit, these circuits can be called trunks. Beware, even though you have a PBX, the Telco may still call these 'lines' (even though your PBX considers them trunks). See also Trunk and Line Side.

Two-way DID trunk- An ISDN PRI equipped for direct inward dialing. Most PBX trunks are related to a given phone number, either alone or as part of a hunt group. In the case of a "normal" (i.e. analog) DID Trunk, a group of phone numbers are associated with that DID trunk (or group of trunks) and incoming calls include the DID number, so the PBX can route that call to the correct DID extension. These are one-way (i.e. inward only) trunks. This is exactly how ISDN PRI functions, with the DID information coming in over the D channel. There is a big difference between a normal DID Trunk and a Two-way DID trunk over ISDN PRI. For one thing, ISDN PRI is digital. Another distinction is that you cannot dial out over a true DID trunk, while you can dial out over a PRI (hence the conflicting designation "Two Way Direct Inward Dialing Trunk").

Two Wire – A circuit path where only a single pair of wires is used. A hybrid is used to convert from two wire to four wire circuits. No hybrid is perfect, and those used by the phone company can be poor. However, the hybrids in Telos TWO/2101 family are approaching perfection!

U Interface- The interface between the ISDN "BRI" line and the "NT1". ANSII standards for the US call for 2B1Q line coding, which is also used in some other places (though not necessarily). This can be considered the ISDN "phone jack" in the USA & Canada, and is frequently in the form of a RJ-11 or RJ-45 style telephone jack. The center 2 pins are used on these jacks.

V.35- A serial data interface/connector pin out for synchronous data. V.35 uses balanced signal and data lines. The Zephyr/Zephyr Xstream models 9X01, 9X00, support V.35 using part #9812 cable.

Variant- The particular protocol (i.e. National ISDN-1 or ETS 300) running on a specific switch. Not all variants are valid for a specific switch. The switch brand and model plus the variant defines the ISND protocol. Applies to configuring the 2101. See ISDN Protocol.

Virtual ISDN- An alternative to repeaters which uses a local Telco Switch to act as a repeater and which then sends the signal onto another switch which supports ISDN. See also Repeater.

Work Order- See Trouble Ticket

X.21- A serial data interface for synchronous data popular in Europe. X.21 uses balanced data and unbalanced signal lines. The Zephyr models 9X01, and 9X00 support X.21 using part #9822 cable.

Yellow Alarm- An alarm on a T-carrier circuit that is returned by the local equipment if it is in a Red Alarm state. A Red Alarm indicates a loss of inbound signal; a Yellow alarm indicates (indirectly) a loss of outbound signal. See Red Alarm, Blue Alarm and LOS.

A3 USA ISDN Ordering Guide

This section applies to users in Canada & USA and is intended to help you when it comes time to order your ISDN lines. We assume you are familiar with ISDN's basic concepts. If you have not read the ISDN section in the manual, doing so now will help you better understand the information that follows.

If you know who to call and you are ready to provide complete information, the ISDN ordering process can be simple and straightforward. This guide takes you through the process step-by-step. Telos support is always available to assist.

On one ISDN BRI circuit, there are three simultaneous bi-directional channels: two 64kbps "bearer" or "B" channels for the transmission of user information and one 16kbps "data" or "D" channel for call set-up and status communication. That is why this is sometimes called 2B+D service. You can use the B channels, but only the phone companies and equipment manufacturers can usually use the D channel. ISDN comes into your facility on two wires, just like plain old telephone service (POTS).

From the perspective of telephone network routing, each channel appears to be a separate line with it's own number and independent dial-out capabilities. Since each has to be dialed or answered separately, they appear to be "lines" to users also.

The Nx12's internal ISDN interface is used to connect to digital ISDN telephone lines. It easily adapts to the various types of service offered by the range of Central Office (CO) switches installed by telephone companies in the USA and Canada.

Dealing with the Phone Company

As is often the case when broadcasters interface with phone people, the lines of communication can get a little tangled.

The first order of business is to find someone who knows what ISDN is. While your usual account agent will be the normal entry point, you may be talking to a number of phone people before you find one who understands your needs.

Some of the regional Bell companies offer a single point of contact number for switched digital services, while some Telcos use "resellers". Note that in some cases the phone company will need to do a "loop qualification" (line loss test) before they can verify that ISDN will be available at your location.

Details, Details

Protocols

In a perfect world, all ISDN terminal equipment would work with all ISDN lines, without

regard for such arcana as 5ESS, DMS100, CSV/CSD, SPIDs, etc. Unfortunately, this is not always the case.

At their central offices, the telephone companies use either AT&T/Lucent 5ESS, Northern Telecom DMS100, or Siemens EWSD switching equipment. While each will work with the Nx12, there are some differences which need to be taken into account when lines are ordered and used. Each has a "protocol" – the language the user equipment and the telephone network use to converse (on the D channel) for setting up calls and the like.

There is a standard protocol that all switches may provide, called National ISDN-1 (NI-1). This protocol was standardized and specified by Bellcore, the technical lab jointly owned by the phone companies. However, both AT&T and Northern Telecom had versions of ISDN which predated the NI-1 standard and some switches have not been upgraded to the new format.

There is also a newer NI-2 standard, but it is designed to be compatible with NI-1 for all of the basic functions, so this is also acceptable.

SPIDs
Service Profile Identification (SPID) numbers are required in all but one of the AT&T protocols. These numbers are given to the user by the phone company and must be entered correctly into the Nx12 in order for the connection to function. SPIDs usually consist of the phone number plus a few prefix or suffix digits. There is frequent confusion between phone numbers and SPIDs, even among Telco personnel. While the SPID frequently includes the corresponding phone number, this is not necessarily the case.

The intention of the SPID is to allow the Telco equipment to automatically adapt to various user requirements by sensing different SPIDs from each type or configuration of user terminal. None of this matters with our application, but we must enter the SPIDs nevertheless. Most of the telephone companies have agreed to implement a standardized SPID for new installations. The standard is area code+phone number+0101 (XXXYYYZZZZ0101).

Your Telco service representative should give you one SPID for each B channel you order. Don't let the phone company installer leave without providing you with the phone numbers of your B channels (called Directory Numbers, or DNs) and your SPIDs!

Directory Number (DN)
These are simply your seven digit phone numbers as would be listed in your local telephone directory and will need to be entered into your TWOx12 (in rare cases you may need to enter the full 10-digit number). Normally each ISDN BRI circuit will have two DNs and two SPIDs. Each DN corresponds to a given SPID and when you enter this information you must enter them to reflect this relationship. You may also hear of a Listed Directory Number or LDN. This is simply the primary number for lines in a hunt group (see below). The LDN need not be entered in the unit, unless it is also one of your DNs.

In some cases, lines configured to hunt may deliver all calls to a single DN number. In this case you must enter that DN for each of the DNs on each of the hunting channels

CSD and CSV
Recall that each ISDN BRI has two possible B channels. It is possible to order a line with one or both of the B channels enabled – and each may be enabled for voice and/or data use. Phone terminology for the class of service is CSV for Circuit Switched Voice and CSD for Circuit Switched Data. (In contrast to PSD, Packet Switched Data, which is possible, but irrelevant to our needs.)

CSD is not required for Telos Nx12 connections. Even though you are sending a stream of data, the phone network knows to translate this back to analog at the far end due to the call setup message sent on the D channel when the call is made. If this ISDN BRI circuit will also be **used**

with an audio codec such as the Zephyr or Zephyr Xstream, then CSD should be ordered as it is required for audio codec calls.

CSV is for interworking with voice phone service and allows ISDN to interwork with analog phone lines and phones.

You will be ordering one or more ISDN BRI 2B+D circuits with CSV (required) and CSD (optional) on each B channel.

NT1s
The ISDN standard specifies two reference points, the U and the S interfaces. The U is the single-pair bare copper from the Telco CO. A device called a "Network Termination, Type 1" converts this to the two-pair S interface.

In Europe, Asia, South America, and most of the world the NT1 is always provided by the phone company. Therefore, Telos TWOx12s shipped outside the USA & Canada have the S interface. In the USA & Canada the NT1 must be provided by the user, and is therefore built-in to the TWOx12's ISDN terminal adapter. In this case, the "U" (RJ-11) interface is provided.

Terminals and Terminal Types
Any equipment connected to an ISDN line is a 'terminal' – whether phone, computer, or Hybrid. Point-to-point lines support one terminal, while multipoint lines can have up to eight in some applications.

"Terminal Type" is a parameter sometimes requested by the phone people. The appropriate value for the Nx12 varies depending upon protocol and is included on the order forms.

IOC Capability Packages (provisioning package)
More and more Telcos are using ISDN Ordering Code (IOC) capability packages for ISDN ordering (National ISDN protocol only). If your telephone company uses these you need only tell them you need IOC package 'S' and an RJ-11 style (SJA48) jack. We still recommend that you fax them these forms as well.

Hunt Groups
Just as with analog lines, ISDN BRI lines can be placed in a 'hunt group' where a single number can be given to callers and the calls will 'roll over' from busy channels to idle channels. This is sometimes called ISG (incoming service grouping) or rollover lines. This would usually be a requirement for an on-air phone system. You may choose to exclude 1 or 2 channels from the hunt group (i.e. the Hot line and the Warm line). Be sure to specify to your Telco how you expect this to work.

FAXABLE ISDN ORDER FORM (for the USA & Canada)

The following form can be used to place orders for ISDN lines in the USA & Canada. Complete the top portion of the form and send all three pages to the phone company. We suggest that you keep the form handy and show it to the installer when he or she puts in your line and ask the installer to verify with the switch programmer (at the central office) that your line is configured as ordered.

You only need to send the first page if the Telco uses IOC (ISDN Order Code Capability) packages for ISDN specification.

ISDN BRI LINE ORDER

USA & Canada
Telos Nx12

To: Telephone Company

Attention: _____

From

Company: _____

Address: _____

City/State/ZIP: _____

Phone: _____

Contact:_____

Location for line

Company: _____

Address: _____

City/State/ZIP: _____

Phone: _____

Contact:_____

Date needed: _____ Termination date: _____

Long distance carrier: _____

Number of BRI circuits required: _____

We request ISDN BRI circuits for use with the Telos Nx12 digital telephone system. This device interfaces studio audio equipment to ISDN. It requires Circuit Switched Voice (CSV). We may also use this line for other equipment that makes calls using Circuit Switched Data (CSD) service. Please advise us if there is a cost penalty for having both CSD and CSV.

If you use IOC (ISDN Order Code Capability) packages, please use Capability Package 'S' . If you do not use IOCs, use the information on the pages that follow.

The Nx12 has an integral BRI terminal adapter which supports these protocols:

- AT&T 5ESS: Custom Point-to-Point (5E4.2 or later), National ISDN-1 or 2
- Northern Telecom DMS-100: Functional (PVC1), National ISDN-1 or 2 (PVC2)
- Siemens EWSD: National ISDN-1 or 2

(AT&T Point-to-Multipoint (Custom) is not supported by the Telos Nx12)

Please let us know which protocol you will provide and the switch. We will provide the NT1 and need a U interface with 2B1Q line coding on a standard, six-pin/4-conductor RJ11-style modular jack (USOC SJA48), of which only the center two conductors will be used.

Hunt Groups (please give us the option specified below)**:**

☐ Please put all Directory Numbers for all BRI's in a single hunt group

☐ Please create a hunt group with all Directory Numbers except (*Specify*) _____

☐ No hunt groups

You may call the manufacturer of the Nx12, Telos Systems, at +1 (216) 241-7225 for any additional required information about ISDN compatibility. Ask for ISDN Customer Support.

PROTOCOL: National ISDN-1

From AT&T 5ESS, Northern Telecom DMS100, and Siemens EWSD.

If you use IOC Capability Packages, please use Capability Package 'S' unless otherwise specified.

CO Values

- Line Type: National ISDN-1
- Bearer Service: CSD/CSV on both channels (CSV is required; CSD may be omitted if requested by user)
- TEI: One dynamic per number
- Terminal Type: Terminal Type A (Voice Terminal Mode)
- 10XXXXX Long Distance Access: Yes

Turn off features such as; packet mode data, multiple call appearances, Electronic Key Telephone Sets (EKTS), shared directory numbers, accept special type of number, intercom groups, network resource selector (modem pools), message waiting, hunting, interLata competition, etc.

Note for AT&T/Lucent 5ESS on loop extended lines: REX test must be disabled for this line using one of the following: ASDFI266 (Prohibit REX on BRI lines per office) or ASDFI267 (Prohibit REX on BRI lines per line)

Give us:

- Two SPID numbers, depending upon number of active B channels
- Two Directory Numbers

PROTOCOL: Northern Telecom DMS100 "Functional" (Custom, PVC1)

Northern Telecom DMS100 switches BCS 31 and above.

CO Values

- Line Type: Basic Rate, Custom Functional
- EKTS: No
- Call Appearance Handling: No
- Non-Initializing Terminal: No
- Circuit Switched Service: Yes
- Packet Switched Service: No
- TEI: Dynamic
- Bearer Service: CSD/CSV on both channels (CSV is required; CSD may be omitted if requested by user)
- 10XXXXX Long Distance Access: Yes

Give us:

- Two SPID numbers, depending upon number of active B channels
- Two Directory Numbers

PROTOCOL: AT&T Point-to-Point (Custom)

Software version 5E4.2 and above.

CO Values

- Line Type (DSL class): Custom Point-to-Point (PTP)
- B1 Service: On Demand (DMD)
- B2 Service: On Demand (DMD)
- Maximum B Channels (MaxChan): 2
- CSV Channels: Any
- Number of CSV calls: 2
- CSD Channels: Any (CSV is required; CSD may be omitted if requested by user)
- Number of CSD calls: 2
- Terminal Type: A
- Number Display: No
- Call Appearance Pref:Idle
- 10XXXXX Long Distance Prefix: Yes

Turn off features such as; packet mode data, multiple call appearances, Electronic Key Telephone Sets (EKTS), shared directory numbers, accept special type of number, intercom groups, network resource selector (modem pools), message waiting, hunting, interLata competition, etc.

Note for AT&T/Lucent 5ESS on loop extended lines: REX test must be disabled for this line using one of the following: ASDFI266 (Prohibit REX on BRI lines per office) or ASDFI267 (Prohibit REX on BRI lines per line)

Give us:

- One Directory Number

A4 Euro-ISDN Ordering Guide

This section applies to users in Europe and other countries where the Euro-ISDN ETS 300 standard is supported (much of South America, Mexico, etc). Since there are virtually no options when ordering Euro ISDN this section is probably not needed, however we thought we could help out by summarizing what you will require.

If you live in a country outside of North America which does not support Euro-ISDN, your local Telos Systems representative should be able to provide you with information about ordering (and using) ISDN in your area. We assume you have read the description of ISDN in section 2.3 and are familiar with ISDN's basic concepts. If you have not read that section, doing so now will help you better understand the information that follows.

On one ISDN BRI circuit, there are three simultaneous bi-directional channels: two 64kbps "bearer" or "B" channels for the transmission of user information and one 16kbps "data" or "D" channel for call set-up and status communication. That is why this is sometimes called 2B+D service. You can use the B channels, but only the phone companies and equipment manufacturers can usually use the D channel. ISDN comes into your facility on two wires, just like plain old telephone service (POTS).

From the perspective of telephone network routing, each channel appears to be a separate line with it's own number and independent dial-out capabilities. Since each has to be dialed or answered separately, they appear to be "lines" to users also. We refer to a B channel as a "line" on the TWOx12's menus and LEDs.

Details, Details

Names
We've seen Basic Rate Interface ISDN called by a number of names. The term "S0" is sometimes used as a technical term for ISDN BRI. Of course the term "BRI" is also used, mostly by Telecom technicians. The most common term we've seen in Europe is "ISDN-2".

Multiple Subscriber Numbers (MSN)
These are simply your phone numbers. The number of digits will vary depending on your area. Normally each ISDN BRI circuit will have three MSNs. You will be using only two of them for each ISDN BRI circuit, one for each B channel. The LDN need not necessarily be entered in the unit (although it would be if it is also one of your MSN's).

CSD and CSV

Recall that each ISDN BRI has two possible B channels. It is possible to order a line with one or both of the B channels enabled – and each may be enabled for voice and/or data use. Phone terminology for the class of service is CSV for Circuit Switched Voice and CSD for Circuit Switched Data. (In contrast to PSD, Packet Switched Data, which is possible, but irrelevant to our needs.)

CSD is not required for Telos Nx12 connections. Even though you are sending a stream of data, the phone network knows to translate this back to analog at the far end due to the call setup message sent on the D channel when the call is made. If this ISDN BRI circuit will also be used with an audio codec such as the Zephyr or Zephyr Xstream then CSD should be ordered as it is required for audio codec calls.

CSV is for interworking with voice phone service and allows ISDN to interwork with analog phone lines and phones.

You will be ordering one or more ISDN BRI 2B+D circuits with CSV (required) and CSD (optional) on each B channel. Euro ISDN normally provides both CSV and CSD which is acceptable.

NT1s
The ISDN standard specifies two reference points, the "U" and the "S" interfaces. The U is the single-pair bare copper from the Telco CO. A device called a "Network Termination, Type 1" converts this to the two-pair S interface.

In Europe, Asia, South America, and most of the world the NT1 is always provided by the phone company, and only the S interface may be on user equipment. If you plan on taking your TWOx12 to Canada or North America you will need to rent or purchase an external NT1 for use while in those countries. One rental source is Silver Lake. See their website at www.silverlakeaudio.com .

Hunt Groups
Just as with analog lines, ISDN BRI lines can be placed in a "hunt group" where a single number can be given to callers and the calls will "roll over" from busy channels to idle channels. This is sometimes referred to as ISG (incoming service grouping) or rollover lines. This would usually be a requirement for an on-air phone system. You may choose to exclude 1 or 2 channels from the hunt group (ie the Hot line and the Warm line). Be sure to specify to your Telco how you expect this to work.

Line Deactivation
The Euro ISDN standard allows ISDN BRI lines to be deactivated by the Telecom when not in use. It is impossible to distinguish a deactivated line from a line which is defective and out of service. If your application is critical you may wish to request that the Telecom disable line deactivation.

FAXABLE ISDN ORDER FORM (for Euro ETS 300)

The following form may be used to place orders for ISDN lines in countries using the ETS 300 protocol. These should give the Telecom all of the information they need.

ISDN BRI LINE ORDER

ETS-300 Euro-ISDN
Telos Nx12

To: Telephone Company

Attention: _____

From

Company: _____

Address: _____

City/State/ZIP: _____

Phone: _____

Contact:_____

Location for line

Company: _____

Address: _____

City/State/ZIP: _____

Phone: _____

Contact:_____

Date needed: _____ Termination date: _____

Number of BRI circuits required: _____

We request ISDN BRI (ISDN-2, S0) circuits for use with the Telos Nx12 digital telephone system. This device interfaces studio audio equipment to ISDN. It requires Circuit Switched Voice (CSV).

The Nx12 has an integral BRI terminal adapter which supports the ETS-300 (Net3) protocol.

Hunt Groups (please give us the option specified below)**:**

☐ Please put all MSNs for all BRI's in a single hunt (ISG or Rollover) group

☐ Please create a hunt group with all MSNs except (*Specify*) _____

☐ No hunt groups

Line Deactivation (please provide the option specified below)**:**

☐ Our application is critical and we need to ensure that lines are working at all times. Please set the switch parameter "Deactivation Allow" to "NO".

☐ We would prefer to know if lines are working at all times. Therefore we request that you set the parameter "Deactivation Allow" to "NO" if this is possible.

☐ Line deactivation is acceptable. Please provide your default setting for this parameter.

You may call the manufacturer of the Nx12, Telos Systems, at +1 (216) 241-7225 for any additional information about ISDN compatibility. Ask for ISDN Customer Support.